

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Automotive Audio

Moving Audio Into an Era of AI and SDV

By Roger Shively

Integrated Audio Development Frameworks

By Philipp Paul Klose

Harmonizing the Future

Crafting Immersive Driving Experiences for Electric Vehicles

By Jasper de Kruiff, David Speith, and Lukas Joas

Car Audio Tuning and Upgrading

The Crucial Question

By Ken Ward

Practical Test & Measurement

Measuring MEMS Microspeakers

By Steve Temme (Listen) and Michael Ricci (xMEMS)



**Simulation Techniques
Dimensional Reduction for
Vibroacoustical Simulations**
By René Christensen

**Multiphysics Simulation
Electrovibroacoustic Analysis of a
Balanced Armature Transducer**
By Jinlan Huang (COMSOL)

**You Can DIY!
A Low-Feedback Triode
Amplifier With Zero Output
Impedance – Part 2**
By Tim Mellow & Peter Farrow

**Hollow-State Electronics
Classic Hi-Fi Before 1950**
By Richard Honeycutt





The new performance PCIe Soundcards.

BY  **DIGIGRAM**

THE WORLD'S LEADING
SOUNDCARD MANUFACTURER FOR

Installation

Broadcast

Audio Measurement

Emergency Systems

ALP-X Series

ALP-222e

ALP-442e

ALP-882e

- 2/4/8 Analog
+ AES/EBU I/Os
- Available with 2/4/8
Microphone inputs
- Headphone output



- Zero Latency
DSP mixer
- Digigram Sound Quality

- Wordclock I/O and
built-in GPIOs
- Drivers for
Windows & LINUX
- Ideal fit for slim form
factor integrated
systems.

Designed for reliable, robust and effective systems in constraint spaces. Working under Windows and Linux environments or with low frequencies.



ALP-Dante



Now with AES67
Dante
SPOKEN HERE

64 x 64 redundant channels

synthax
PRO AUDIO DISTRIBUTION
FOR THE AMERICAS

GET A FREE CONSULTATION
+1 754-206-4220

digigram.synthax.com

SPEED MATTERS

ACCELERATE YOUR IN-CAR AUDIO MEASUREMENTS WITH SOUNDCHECK®



From the R&D lab to end-of-line tests, SoundCheck® is fast, flexible and cost-effective.

- Power-averaged frequency response / tuning
- Max SPL
- Impulsive distortion
- Spatial audio
- Hands-free communications
- CarPlay and Android auto
- End of line test
- Road noise and ANC testing



Learn more: www.listeninc.com/automotive



June 2024

ISSN 1548-0628

www.audioxpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by KCK Media Corp., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
E-mail: custserv@audioxpress.com
Internet: www.audioxpress.com

Postmaster: Send address changes to:
audioXpress
P.O. Box 417
Chase City, VA 23924, US

Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence, suggestions, and contributed articles to:
audioXpress, Editorial Department
P.O. Box 417
Chase City, VA 23924, US
E-mail: editor@audioxpress.com

Legal Notice:
Each design published in audioXpress is the intellectual property of its author and is offered to readers for their personal use only.
Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2024
Printed in the US



Different Listening Perspectives

I recently attended a very successful AXPONA (Audio Expo North America) 2024 edition in Chicago, IL, where I was pleasantly surprised with the vibrant showcase of audio systems that filled the largest number of rooms in the history of this show. Hundreds of rooms with lots of innovative speakers to discover (some doing their premiere here, instead of waiting for the High End show in Munich), but also lots of new electronics, including many network audio streamers, which were clearly the most popular audio source, followed in a distant second by vinyl and CD—which are both mandatory as many attendees bring their own physical media to listen to for reference. These high-end streamers are now a core hifi component because they allow an aggregation of streaming services, enabling sessions to freely pick tracks from the Tidal, Qobuz, or Apple Music, all of which now offer lossless streaming options—or local NAS and storage devices for Hi-Res PCM and DSD files.

On the second day of the show, I noticed that the hotel Wi-Fi network that had worked reliably until then was suddenly absent. Apparently, the excellent facilities of the Renaissance Schaumburg Hotel & Convention Center were no match for the hundreds of simultaneous FLAC streams demanded by an audio show. Or maybe it could just have been a local outage in Schaumburg’s telecom service—I did not inquire, since services resumed as I was reaching the hotel convention area again.

Using streaming services to quickly find a particular song has been a practice at audio shows for quite some time—but was previously not something to brag about until lossless streaming became available. In general, there was a preference for playing files from a local server, enabling access to uncompressed masters and high-resolution files. Now, with the availability of “CD-quality” 16-bit files streamed in lossless flavors it seems everyone just accepts it, and no one seems to bother hiding it from the attendees—also because in most cases that’s what they already do at home.

AXPONA was a “stereo-only” showcase. There’s absolutely no interest in playing the “Spatial Audio” mixes that are available through the same services. And for some reason, streaming lossless FLAC of standard CD quality files is considered “hifi” enough, and there’s far more enthusiasm for rediscovering expensive and extremely inconvenient analog alternatives, such as listening to reel-to-reel recordings.

From Chicago I moved to Las Vegas, NV, to attend the National Association of Broadcasters (NAB) annual convention. Every NAB Show is a vibrant showcase of the latest cutting-edge technology for content production, from TV and radio to movie production, to “the Creator Economy,” which these days means streaming, podcasting, and any content distributed directly on the web and social media.

In these environments—where a massive percentage of today’s content is generated—24-bit digital audio at a minimum of 48kHz is standard for hybrid AoIP/cloud infrastructures, where codecs are as common as the number of “native” file formats. Audio is mostly recorded at the industry standard 24-bit/96kHz using Broadcast WAV files, and absolutely no one calls this “hi-res.” It’s just audio.

In post, mixing 5.1 surround and Dolby Atmos is now the default for any type of scripted content. This means that productions expect to use music already mixed for immersive formats.

The NAB Show is also where the podcasting community shops for technology. And podcasting is increasingly a sophisticated multiplatform environment with high production values, where the use of artificial intelligence for things such as editing and transcription is becoming very common, and mixing for Spatial Audio is also becoming common practice.

As this issue of *audioXpress* focuses on Automotive Audio, an extremely important listening environment that is definitely not designed for “stereo,” this also puts things into a totally different perspective. According to Edison Research’ latest Share of Ear report, 53% of today’s 13-34-year-olds are listening to radio every single day, mostly in the car. Also according to Edison Research, the US podcast audience today approaches 100 million weekly listeners.

Automotive audio today is designed for immersive experiences with access to “on-demand” streamed forms of contents, where digital-native “hi-res” immersive podcasting will thrive.

J. Martins
Editor-in-Chief

The Team

Publisher:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Tori Zienka
Editor-in-Chief:	João Martins	Advertising Coordinator:	Heather Childrey
Technical Editor:	Jan Didden		

Regular Contributors: Ken Bird, Bruce Brown, Bill Christie, Bob Cordell, Joseph D’Appolito, Dimitri Danyuk, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, David Logvin, Ward Maas, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Michael Steffes, Nora Wong, Stuart Yaniger, Ethan Winer

OUR NETWORK



SUPPORTING COMPANIES

Clockworks Signal Processing, LLC	35	NTi Audio, Inc.	65
COMSOL, Inc.	9	Parts Express International, Inc.	68
Dolby Laboratories	23	QuantAsylum USA, LLC	41
DSP Concepts	17	Redco Audio	55
Elytone Electronic Co., Ltd.	61	SB Acoustics	45
FlextechAKT	21	Sonical	31
HEAD Acoustics GmbH	25	Synthax, Inc.	2
Impulse Audio Lab GmbH	31	Triad Magnetics	67
KAB Electro-Acoustics	21	Tymphany Acoustic Technology (UK), Ltd.	39
Listen, Inc.	3	Warkwyn	55
Menlo Scientific, Ltd.	51		

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706) to reserve your own ad space in the next issue of the magazine.

The Audio Voice

Sign up now for our **FREE** weekly e-newsletter

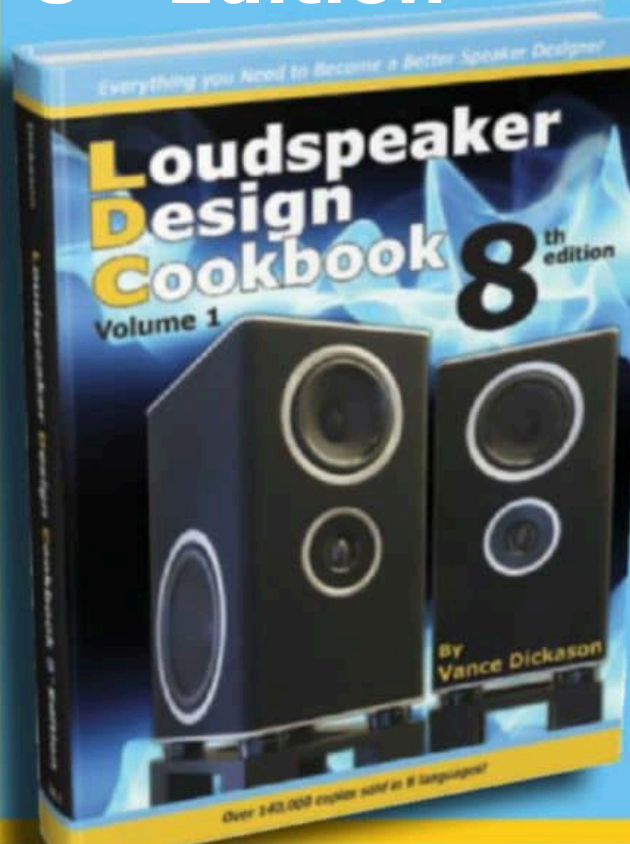


Weekly Newsletter for the Audio Industry, Professional Designers and Audio Enthusiasts
Product Design | Audio Electronics | Acoustics | DIY | Innovations in Audio



Loudspeaker Design Cookbook

8th Edition



Learn to Design Professional Sounding Loudspeakers for:

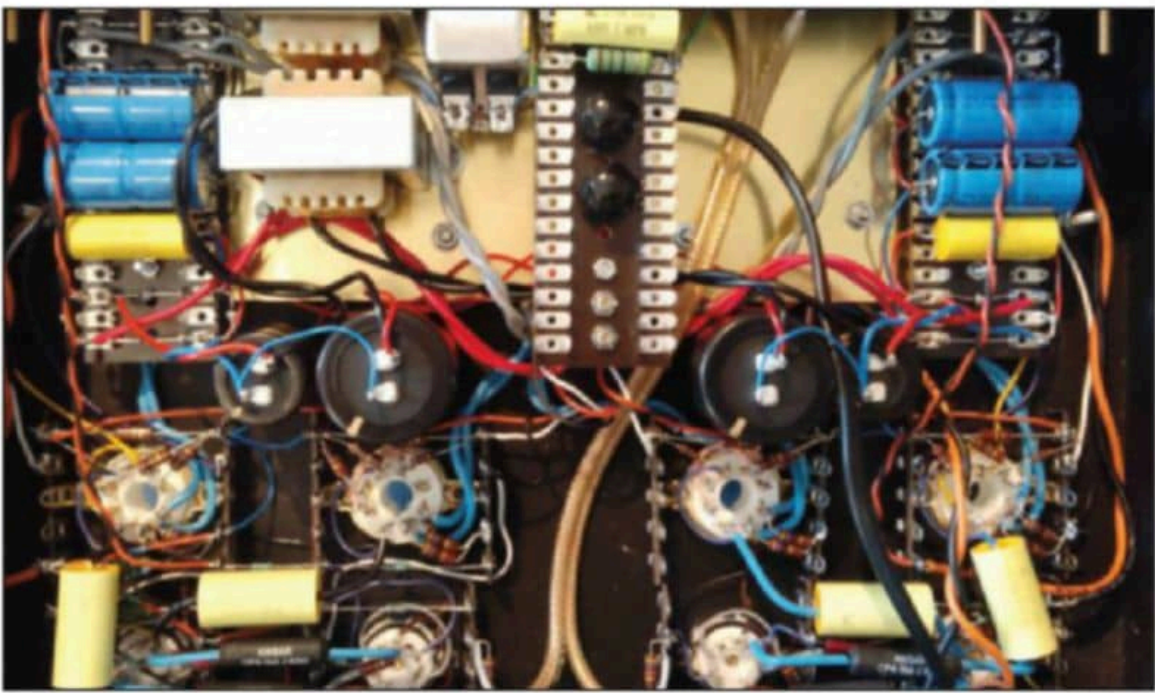
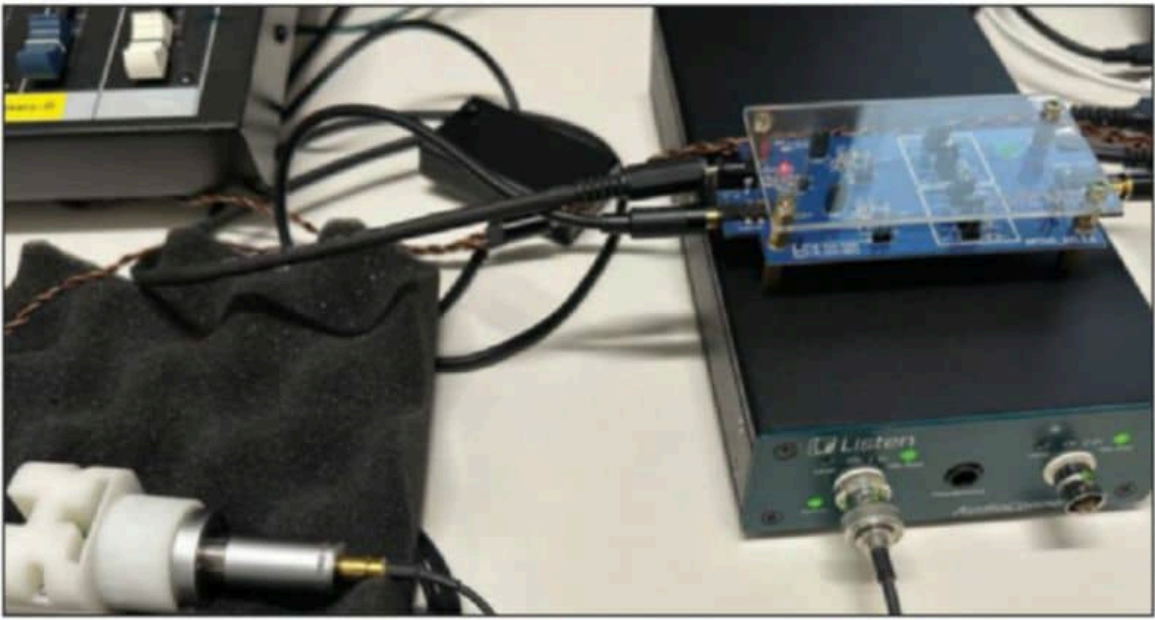
- Residential • Automotive • Home Theater

Vance Dickason's top-selling *Loudspeaker Design Cookbook* is an essential must-read for serious speaker designers, audio engineers, and anyone looking to master speaker-building technology. With more than 140,000 copies in print, author Vance Dickason's *Loudspeaker Design Cookbook* is not only regarded as the loudspeaker engineering "bible," but the #1 resource on the construction of high-quality loudspeaker systems for two-channel, home theater, pro sound PA speakers, and studio monitors, as well as car audio applications.

For the First Time Ever...
Available in Full Color
and a Hardcover Collector's Edition

CC-Webshop.com/LDC8





● Features

36 **Using Multiphysics Simulation for an Electrovibroacoustic Analysis of a Balanced Armature Transducer**

By Jinlan Huang

This article explains how multiphysics modeling and simulation can be used to analyze electroacoustic transducers. Using a balanced armature (BA) transducer as an example, the analysis combines the electromagnetic field with structural vibrations and acoustic radiation using multiphysics couplings.

46 **Dimensional Reduction for Vibroacoustical Simulations**

By René Christensen

Computational demands can be very high when doing simulations involving coupled physics, as is often the case for vibroacoustical transducers. This article looks at how to reduce the computational demands and associated costs utilizing symmetry schemes and dimensional reduction techniques, and the potential associated issues.

52 **Measuring MEMS Microspeakers**

By Steve Temme and Michael Ricci

MEMS microspeakers are now commercially available and seeing fast evolution and adoption in true wireless stereo (TWS) earbuds and in-ear monitors (IEM). With the emergence of these small form-factor transducers comes the need to address how to measure its performance and ensure product quality.

56 **A Low-Feedback Triode Amplifier with Zero Output Impedance Part 2 — Balanced Input Amplifier Design, Performance, and Detailed Calculations**

By Tim Mellow and Peter Farrow

The first part of this article introduced the Zero Output Impedance amplifier concept and discussed the design for a single-ended input version. In this second part, the design is extended to a balanced input configuration. Also, performance measurement figures are discussed, and two Appendices give design calculations for both configurations.

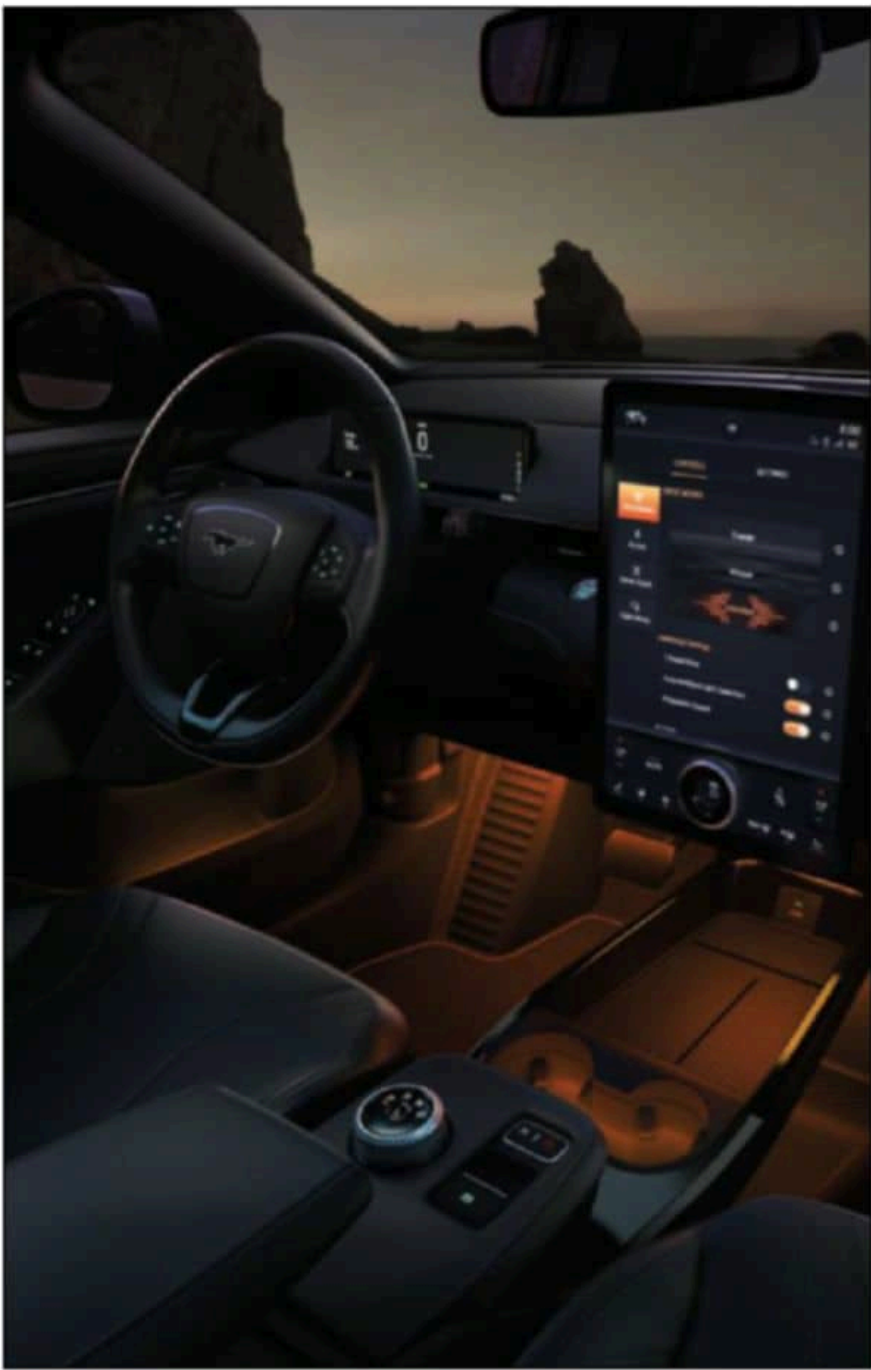
● Columns

HOLLOW-STATE ELECTRONICS

62 **Classic Hollow-State Hi-Fi Before 1950**

By Richard Honeycutt

The term high fidelity (later shortened to “hi-fidelity” and eventually just “hi-fi”) was used as a marketing label in the 1950s to promote the difference in sound reproduction from the technologies available in previous decades. Yet, high fidelity was not exactly a new concept in home music playback systems.



● Automotive Audio

8 Moving Audio Into an Era of AI and SDV

By Roger Shively
Automotive audio is shifting from hardware-dependent development as the industry embraces software-defined vehicles (SDVs) and Artificial Intelligence (AI) finds a practical role. How is this changing audio systems in the car, and what changes in the way audio systems and infotainment are designed?

18 Integrated Audio Development Frameworks

By Philipp Paul Klose
Developers and designers are familiar with a number of audio frameworks that enable more time for actual innovation and less time to write code, debug, and test hardware deployment. New and existing integrated development environments (IDE) are essential as automotive audio evolves in tandem with software-defined vehicles.

26 Harmonizing the Future
Crafting Immersive Driving Experiences for Electric Vehicles

By Jasper de Kruiff, David Speith, and Lukas Joas
Vehicle sound design is essential to ensure that the transition from combustion engines to electric power doesn't compromise the thrilling essence of driving. For this purpose, Impulse Audio Lab created EVx, a specialized software suite for the creation of dynamic and complex sounds. Haptic seats developed by Icon Incar add synchronized tactile feedback and create a compelling synergy of audio and haptics.

32 Car Audio Tuning and Upgrading
The Crucial Question

By Ken Ward
Aftermarket car audio always needs to be built on top of the audio systems that the manufacturers designed. This provides an opportunity to understand what works and what can be improved. A perspective that is always valuable for any designer.

● Departments

- 4 From the Editor's Desk
- 5 Supporting Companies
- 66 Article Update

● Websites

- www.audioxpress.com
- www.voicecoilmagazine.com
- www.cc-webshop.com
- www.loudspeakerindustrysourcebook.com
- www.circuitcellar.com
- www.linuxgizmos.com

 @audioxp_editor

 audioexpresscommunity

 linkedin.com/company/audioxpress

 audioxpress



Moving Audio Into an Era of AI and SDV

By
Roger Shively

Automotive audio is shifting from hardware-dependent development as the industry embraces software-defined vehicles (SDVs) and Artificial Intelligence (AI) finds a practical role. How is this changing audio systems in the car, and what changes in the way audio systems and infotainment are designed?

With supply issues behind us, used vehicle prices normalized, and supply built back up to expected levels, the automotive industry has fully recovered, albeit in a world of limited demand. In the US, Cox Automotive estimates that 2024 will see 15.6 million units in new car sales, a 2% growth over last year. That is the most normal level of sales in the last five years. It is also estimated that EV sales will increase to become 10% of the US market, stemming from more production output of EVs.

Even with all the positive energy that abounds in the automotive industry and the exciting vibe that existed at CES 2024, it was possible to look at the excitement with a more dissonant feeling than in the past year—a feeling rooted in the real shift from hardware-dependent development in automotive to software dependent development. With the idea of a software-defined vehicle (SDV) becoming mature with application, Artificial Intelligence (AI) also continues to find unique, practical ways for integration into our audio systems.

The Role of AI

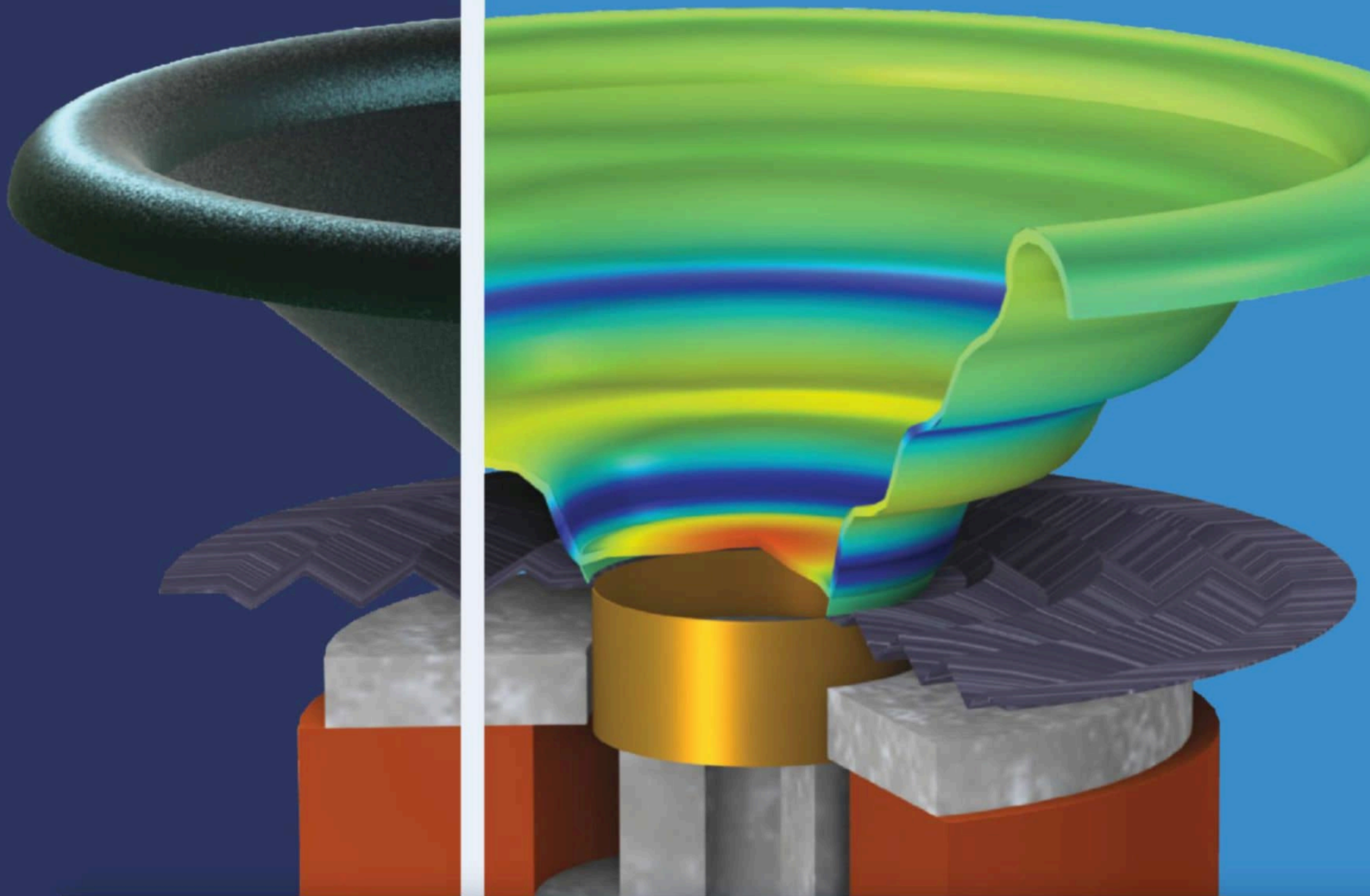
At the start of 2024, it was clear that gone are the sky-high commitments and the lightspeed timelines given for promising new technologies, including AI. High interest rates, tighter capital access, and a growing impatience from investors and customers have changed things. Also, expectations for automotive AI have come back down to less rarefied air.

Most companies at CES and at the SAE World Congress in 2024 showcased automotive AI in lower-stakes applications (e.g., vehicle personalization) rather than in complex functions (e.g., autonomous navigation on busy city streets). The aspirational aspects of products have become more mature and ready for prime time: practicality, less flashiness, and creating greater efficiency in decision-making are now the goals of automotive AI.

Rather than having engineers manually code responses to individual stimuli and edge cases, vehicles may instead learn from examples, like humans do, performing tasks better, rather than striving for perfection.

Shortly after the release of ChatGPT, Ethan Mollick, Associate Professor at the Wharton School of the University of Pennsylvania, wrote in an article for the *Harvard Business Review* in December 2022 that “Until now, AI has primarily been aimed at problems where failure is expensive, not at tasks where occasional failure is cheap and acceptable.”

Chatbots such as ChatGPT and other generative AI tools were fertile ground for the automotive industry this year. A lot is centered around voice assistants, personalization of the in-vehicle experience, and a little on the digital customer experience. Automakers are delivering on the low-hanging fruit, including internal AI tools that help engineers write code and streamline processes, such as research and design, that are invisible to the consumer. Customer service bots, driver and



Take the Lead in Acoustics

with COMSOL Multiphysics®

Multiphysics simulation drives acoustics innovation by providing insight into every design aspect that influences product performance. The ability to account for coupled physics phenomena lets you predict, optimize, and virtually test a design under real-world conditions – even before a first prototype is built.

» comsol.com/feature/acoustics-innovation



AFEELA by Honda: Sony combines AI for ADAS with Cognitive Neural Networks to enhance the gaming experience in Automobiles, using vehicle and driver data to create a gaming creative entertainment community space, so we can bring our digital life into the vehicle with us.

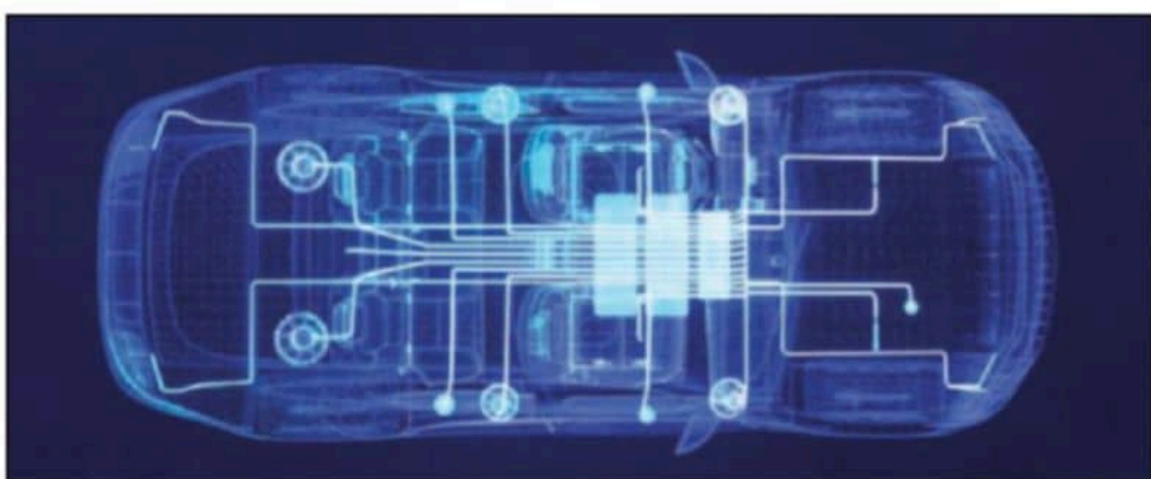
interior monitoring, as well as tools to predict maintenance and repair needs are the new frontier.

Automotive companies and their suppliers are increasingly trying to develop technology that takes advantage of multimodal AI, which enables the developer to provide multiple types and modes of data, and then generate an output for each of them. It is technology that can generate text, audio, and video, while monitoring tire pressure, climate control, and other vehicle features. All of this is part of an industry drive to make vehicles as personalized as possible—to which the audio system will be central.

Software-Defined Vehicles

Even though Danny Shapiro, Nvidia's vice president of automotive, recently told *Automotive News Europe* that "The naming is a little goofy," the SDV has been an industry buzzword for a few years now, and it seems to be on the verge of breaking through with real applications. Compared with previous years, in 2024 Qualcomm refocused its approach to automotive and talked about its Hexo-Software Factory, rather than new chipsets.

Automakers hope that shifting emphasis from hardware to software will cut development costs, strengthen customer loyalty, and increase revenue. However, creating a smartphone-on-wheels



BlackBerry QNX Sound is a development platform for in-vehicle sound that decouples audio and acoustics software from the vehicle hardware. The new BlackBerry QNX Sound platform is expanded with Dirac's full suite of software optimization tools.

requires a complete rethink of vehicle development, new supplier relationships, and an overhaul of corporate culture.

What counts as a "true" SDV is up for debate, but most agree it needs to have a centralized computer system with just a handful of powerful ECUs controlling a range of on-board functions as well as data flow to and from the cloud. Danny Shapiro pointed at his mobile phone, and said "Do you call this a software-defined phone? No, but the principle is the same. It just means the car has a computer that gets updates."

Because the focus now is getting the latest and best software experience, features, and user experience, what used to be a software selection based on the hardware in use is now reversed, as the hardware is dictated by the software. This starts with a future vision, and then the idea of what is needed to achieve this. Following that, then, is the development of a chip, hardware, or the electronic/electric (E/E) architecture, to use the topology. That is a reversal of the traditional model for automakers. It's not only about the chips and the hardware, but also about how the software is developed.

"The promise of the software-defined vehicle is going the same way as the promise of autonomous vehicles. It is difficult," argues Jean-Marie Lapeyre, Capgemini's chief technology and innovation officer covering automotive. Lapeyre's experience in IT and Information Security goes back many years, having worked for GM, Opel, and PSA Group.

During a recent round table discussion in the UK, led by *The Financial Times*, Lapeyre referenced the Gartner "hype cycle," which represents the five key phases of a technology's path to maturity. These are the technology trigger, peak of inflated expectations, trough of disillusionment, slope of enlightenment, and finally the plateau of productivity. He placed SDVs on the downward slide approaching the trough of disillusionment, the low point that follows a height in optimism as a good idea moves forward to its difficult execution. Although this "cycle" isn't technically cyclical, it is highly subjective and isn't scientific in nature, it nevertheless has been used to assess the maturity level of a new technology.

The defense industry does something similar, using technology readiness levels (TRLs) to estimate the maturity of new technologies: basic principles, concept formulation, proof of concept, lab validation, field validation, system level prototype, and qualification. Both the Gartner hype cycle and TRLs can be used to assess how new technologies emerge in a modern, venture capital-based world and to signal when they are viewed as at or near maturity by the industry that is adopting them.

Even if SDVs might have entered a "trough of disillusionment," expectations haven't dimmed despite the recent challenges. Volkswagen Group's troubled CARIAD software division had setbacks that stalled the launch of the new, full-electric Audi Q6 E-tron and the battery-electric Porsche Macan. For Volvo electric vehicles, EX90 flagship SUV was delayed because of challenges encountered while testing "mission critical" software, while deliveries of the EX30 small SUV were stalled for half a month because a crucial software update could only be performed at dealerships rather than over the air. Renault, meanwhile, hopes to halve vehicle development time by reusing software developed

in the cloud and downloaded onto a new “digital chassis,” created with the help of chipmaker Qualcomm and an Android operating system.

Rapid Changes

In 2021, just 2.4% of vehicles on the road could be described as software-defined (mainly Tesla models), according to data from Deloitte. By 2030, “90% or more” will fit that definition, Deloitte predicts.

“Customers have an expectation in terms of what a consumer product is, because they hold a smartphone or a tablet in their hand all the time,” Qualcomm head of automotive Nakul Duggal stated. Qualcomm transformed its expertise in consumer electronics as a chip supplier to the smartphone industry into that of a crucial supplier to the automotive industry.

At CES 2024, Bill Pinnell, Vice President and head of Automotive Software Product Management at Qualcomm, talked about the company’s Hexo-Software Factory solution to address the problem of the current E/E architectures, starting with a challenge and ending with what he felt was a mature solution. The challenge is that there are tens or hundreds of millions of lines of code in a vehicle. When such a level of software needs to be updated, and the latest version of Apple or Android needs to be added—or whatever it is that the consumer and the OEM want to do—then a different software design and development methodology is also needed.

In other industries, particularly in cloud computing, this is nothing new. So, what we call cloud-native development—people writing code that runs in the cloud—refers to micro-services, or service-oriented architecture (SOA). These are very well-established principles, and now we’re seeing their appearance in the automotive industry.

So, when developing software, there is no desire to have to use a particular hardware development board. If there are 200 developers, it would mean 200 development boards, which is unreasonable. The desire is to want thousands of developers that can be approached and used, and they would be developing in the cloud. They’re going to be using modern, containerized technology, such as an open-container initiative (i.e., a container that’s going to run either infotainment or a digital cockpit, or even ADAS functionality). All of that can now be containerized and run in these higher performance compute platforms for which the SDV is really designed.

This becomes more tangible when it is understood how it actually takes place. One of the things Qualcomm does with OEMs and Tier Ones, and other software suppliers, is work with what is called a hexo-software factory.

This is basically a collaborative model, where several parties can basically take code from multiple suppliers, traditional Tier Ones, or modern ODMs that are going to be building hardware, as well as from the OEM itself, because the OEM has its own signature applications that it wants to bring into its car. This is instead of having the more traditional model where there would be a hierarchy of Tiers: Tier Three supplying software and/or hardware to a Tier Two, Tier Two supplying to a Tier One, and then the Tier One working directly with the OEM. Such a model worked for the



Like a smartphone? The Spirit of Mustang in the electric Mustang Mach-E is software-defined and controlled by the driver in the central display, which generates “propulsion” sounds and supports Wireless Apple CarPlay and Android Auto.

lower complexity of applications that have existed for the last 15 years. But now, with all these bits of software and the hundreds of millions of lines of code, they all need to work together.

It’s a model that goes something like this: When you push something into the software factory, you’re using agile methodologies; so, you’re doing product increments, sprints, your user stories, and you’re demonstrating code to the different suppliers and management within the OEM. You’re showing, for instance, the latest HMI feature—it could be something like a new augmented reality (AR) feature on your heads-up display (HUD) that you want to show off to the management team. You’ve worked on your user story, your product team has developed it, but they haven’t completely fleshed it out yet. You demo it, and the management team on that day says something like, “No, we think we need that to be over here, or we want to change this,” or whatever, and the teams will work together to make that happen. But there are dependencies on different suppliers. You don’t have to be responsible for someone else’s code working, but you do have an established test and design methodology, you have test



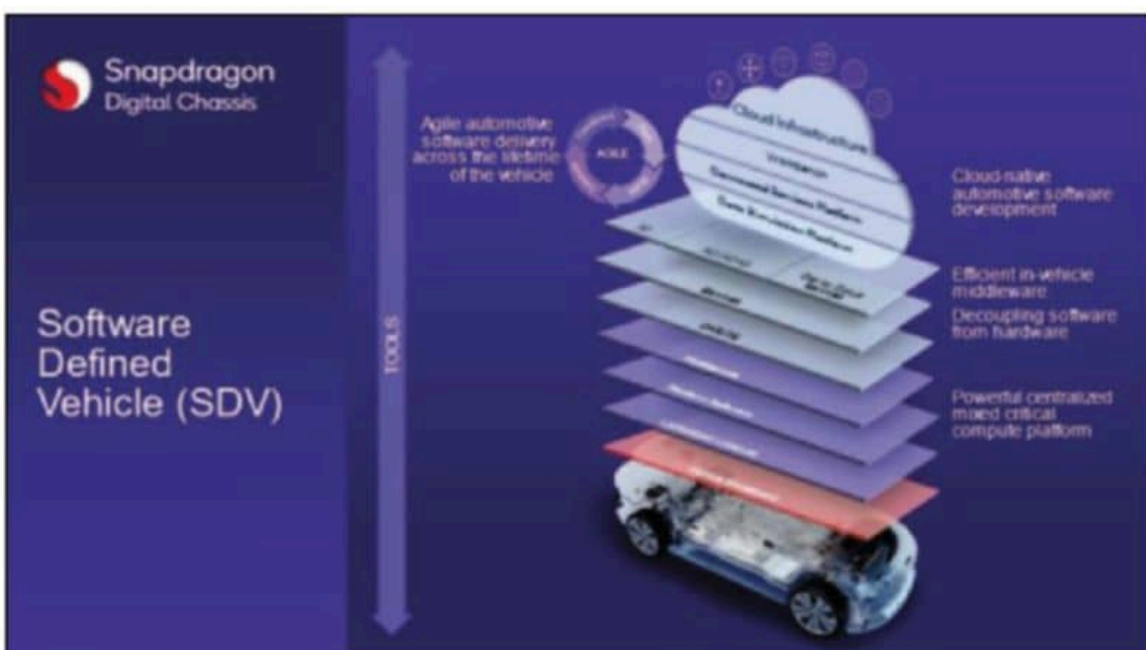
To promote its Dolby Atmos for cars program, Dolby is working closely with the complete automotive ecosystem, from content creators, and streaming services, all the way to Tier One partners. This is making it easier for automakers to offer the full Atmos immersive experience, as recently confirmed with the new Xiaomi SU7 Series, the first ever EV from the consumer company.



Qualcomm joined industry-leading consortiums to accelerate software-defined vehicle (SDV) technology.

plans where you're responsible for your own API working, you've proved it works, and therefore when you test it with someone else's and it doesn't work, you can conclude there must be something wrong on their side, and you all work together in collaboration to resolve the issue.

The algorithms for automotive audio play into this directly, which is why Harman and Bose are not hardware-focused, but rather software-focused. And this is why Dirac and others are looking for a place on platforms such as BlackBerry QNX and DSP Concepts' Audio Weaver. For someone like BlackBerry QNX, which is a close partner with Qualcomm, their goal is to revise the whole architecture for the industry. They need to see teams talking to each other, which is very difficult today. As soon as the platform designer, such as BlackBerry/QNX or DSP Concepts, start to talk about the details of engineering and acoustics with the teams, that's where they lose the real collaborative nature of a platform design. They are connected to each of the individual engineering teams, and when they start to pull back the carpet to look underneath, they find a very segmented architecture that isn't very well connected. Everything becomes an R&D project instead of product development. Today, most of their effort is helping OEMs make the right decisions in the beginning so that the soundscape of the car is a coordinated, holistic, end-to-end



Qualcomm's SDV framework is based on a rich software stack that uses hypervisors for virtualized processing.

design. There is not one war room where all the decision makers are working together to make a holistic design. It is not efficient, it costs more, and it isn't the best for the end customer.

Now, in the middle of 2024, all the 2026 projects are done. The development cycle is following the cycle of the systems-on-chips (SoCs). There will be no more design wins this year. All the request for quote (RFQ) battles were fought in the last 2 years. But it's a good time for design, and the platform designers will work to be ready in the next 2-3 years. Now is a good time to make good decisions. The next chip generations (e.g., Gen 5 of Qualcomm chips) will be able to include Infotainment and ADAS on the same chip, same for Samsung and Nvidia. Audio providers will not be sourced in the future the way they have been in the past. There is more and more software. Just like on smartphones, the hardware is established, and the software is forced into it. That is what will happen for audio in the next car models. Right now, demo platforms that are more open to collaborations is important, to show the eco-system to the OEM. There are OEMs looking at the correct system approach. For example, they have 16 demo car setups for the executives to experience, and if the executive experiences the feature and says "yes," then it gets into the architecture. And they continue to the next, building a connected system architecture.

Progress Is Being Made

The risk of making sweeping software changes within an established automaker is even greater. Starting production on time is hard enough without having to chase down bugs in a new E/E architecture that is running new software from suppliers the automaker has never worked with before. If everything is done at once, development is overloaded, potentially putting the entire schedule at risk. However, those schedules are being upended because automakers know they must take the risk if they want to transform the car into a smart device that offers seamless updates, services, and data-driven intelligence. With the current architecture, it is not easy to implement the services everybody dreams about. It requires a system with a central computer, itself another sweeping change.

Most automakers have now combined the driver display with the central touchscreen in a sort of predecessor to the zonal computer architecture to create a coherent infotainment system with over-the-air (OTA) updates and additional downloadable features. But as is the case with most tech developments in the automotive industry, Tesla is leading the race to create the software-defined vehicle. Chinese startups such as Nio are not far behind, similarly unconstrained by a legacy development culture.

"Our ambition is to become a software-defined company," said Derek de Bono, vice president, software-defined vehicles at Valeo. "Not to the detriment of hardware, but the software will define what hardware we need and what we can do with it," he said at CES this year. "You can design your product, partner with any kind of player, then start right away."

As new partnerships evolved, it became clear that software-focused suppliers had access to data and insights generated by the car's usage that was never previously available. "That is why

CARIAD was founded, because we did not have access,” Dirk Hilgenberg, the former head of CARIAD, said at the 2023 edition of CES. “We wanted to gain access and have our own data-driven development process.” The move backfired when, despite hiring more than 6,000 software specialists, the scale of the task still proved too great for CARIAD. Last year, Hilgenberg was replaced, and the division reportedly was planning to cut 2,000 jobs. CARIAD tried to do everything on its own. Now, the company is more realistic, understanding it needs to work more with existing suppliers. Indeed, software development can be very expensive if you go it alone, with Deloitte estimating that an automaker would need to spend USD \$2.7 billion a year to maintain its own in-house software unit.

Managing relationships with suppliers in the software age isn’t easy, though. The industry needs to move away from “value chain” and toward “value network,” Christian Sobottka, president of automotive for Harman, said at CES this year. “There was a period when certain automakers were trying to apply the old industry model to the new challenge. Over the last two years, some of them have struggled and failed.” A cultural transformation is required. “The cooperation model of the past where the automaker defined everything and commanded a certain set of suppliers is not good to lead us into the future. There is a huge partnership needed,” added Sobottka, formerly CTO at supplier Robert Bosch.

The Software-Defined Vehicle Alliance was launched at CES 2024 this year to push for standards and development methodologies. The idea is that anything developed in terms of software will then easily communicate with the middle and base layers of software in the car, allowing new code to be written and tested in the cloud (known as the “digital twin” method). This would theoretically cut months of model development time and save millions by avoiding “hardware-in-loop” bench-testing in prototype vehicles.

The industry groups were set up out of frustration, seeing the automakers as being stuck in their ways and unable to move. For example, the British chip and software designer ARM created Scalable Open Architecture for Embedded Edge (SOAFEE) in 2022 to give the SDV industry a much-needed push. SOAFEE’s members include Continental, Tata Motors, Geely, Qualcomm, and Harman. In this new supplier relationship, automakers can focus on the software the customer sees, allowing for brand differentiation. Suppliers, meanwhile, can work on the layers below, allowing them to create scale by reusing software. Independent app developers can work on car-specific services with the knowledge that they will be compatible with a range of brands because of a common “abstraction layer,” where instructions can be passed without needing to know the details behind them. APIs can be the software mechanism for accessing the features of a given platform.

You Still Need Chips

Automakers are leaning heavily on chipmakers such as Qualcomm, Nvidia, and Renesas to supply centralized SoCs powerful enough to deal with the increased screens, enhanced graphics, and additional data processing needed. Much like many hardware companies hoping to get into the automotive industry and sell their new designs found that it was easier to partner or merge



Characterized as “a scalable framework for software-defined vehicle experiences” Qualcomm’s Snapdragon Cockpit Platform is a set of processors designed for AI and cloud connectivity to deliver personalization, and safety enhancements to occupants and the automotive ecosystem.

with an existing hardware company, likewise, chipmakers who are new to automotive discovered it was easier not to compete with existing frameworks, and instead to create the platform for the frameworks to sit on.

Although tech companies are pushing their way into a space almost exclusively occupied by traditional automotive suppliers, several established players are also winning substantial business. Harman last year won what Sobottka described as the company’s largest automotive order ever for a centralized computing system for a single company worth “multi-billion dollars.” He did not name the company. Mobileye last year booked orders of USD \$7.4 billion. The former Israeli startup, now owned by Intel, has shifted strategy from offering a “black box” ADAS product to letting automakers tune the final product via its DXP operating system.

Whether a return on the investment pouring into SDVs will be seen remains an open question, as suppliers and automakers investigate which content and services customers will be willing to pay extra for. What won’t necessarily work, for example, is asking



At CES 2024, Nakul Duggal, SVP & GM of Automotive & Cloud Computing, Qualcomm Technologies spoke about the process of “Coding the Car” describing the path to Software-Defined Vehicles.



Harman is pushing automotive connectivity to create compelling in-vehicle experiences and more amplified in-cabin experiences for drivers and passengers alike. Its Ready Connect portfolio also reduces time to market and engineering efforts for OEMs.

customers to pay to unlock embedded hardware, a problem that BMW ran into recently after promising to drop its idea of a heated-seat subscription. The concept of the “App Store” for automotive features is something that is appealing for automakers but is not easy for consumers.

Consumers would gladly accept the idea if it meant making the car cheaper and allowing them to pay only for things they want or need. But consumers are also suspicious and tend to believe they are being primed for upselling. Automakers have seen the value of selling audio and video streaming, but even after experiencing a compelling audio demonstration in a car, consumers might find it hard to pay extra for abstract things such as algorithms. It will be in the hands of the audio brands to make it compelling and in the hands of the automakers to keep the consumer aware and interested in the something new that comes along.

There is also the issue of regional regulations and taxes. Where will the “App Store” reside, in a tax-haven cloud? Do consumers pay VAT for these apps in addition to the local sales tax? This is a similar issue for buying a car in the European Union. For example, the same car purchased in Germany might cost twice as much in another EU country whose GDP is a fraction of Germany’s due



Harman is leading an industry-wide Software Defined Vehicle effort to build a vendor-agnostic, open-source ecosystem. Transitioning to a modern software architecture in cars is “costly, painful, and intense,” says Harman President of Automotive Christian Sobottka.

to local special taxes and higher VAT, which is in addition to the tax on the car.

Similar regional issues exist for data security. One automaker that uses Google as its OS, ran into a problem with at least one prominent Middle East country that would not allow any data from its home country to be transferred to a server located outside the country. A solution was eventually found, but it was not the impossible swapping of hardware, it was working with Google to rewrite the code that was used in that country so the data was isolated to the local Google servers. The joy carmakers might have in the prospect of “reselling” the same car with a software update in year or two, is complicated by not being able to sell the “same” car everywhere.

Yet, the drive for SDVs is strong, and it will have an impact on how audio features are delivered to vehicles and how they are developed. “The constant is that you need more computing power. There’s a direct link between the amount of compute in the car and how safe that car can be,” and how many additional features can be provided, Danny Shapiro, head of automotive at Nvidia, told the software-defined vehicle panel at CES 2024.

Future-proofing a car with oversized chips is a possibility, but the desire to improve E/E architectures with the latest equipment for subsequent models will erase some of the cost-saving from repeatability. “Nothing lasts forever,” Fisker CEO Henrik Fisker mentioned. “If that was the case, you would still be running around with your first iPhone.” The improvements in ADAS and the potential for downloadable, hands-free driving as promising revenue streams will drive a computer power race.

Automotive audio will be going along for the ride in the computer power race. Audio software platforms, such as Blackberry QNX and DSP Concepts will be integral in providing the audio solution to a SDV. Chipmakers, such as Qualcomm, Analog Devices, NXP, and Texas Instruments (TI), are still where low latency requirements will be met, and the platform will enable the software, such as ANC and Immersive Audio, to work on the chip level and then communicate up to the system level. Those platforms will find their comfortable space in the larger software framework with which they will be collaborating, even if it is Apple or Google in the end. However vast the problems in creating the SDV are, and whatever the consumer’s real desire for the products may be, the race to solve them is becoming more intense as Tesla and several Chinese startups work through the same issues without the drag of legacy cultures. Right now, automakers are racing to catch up with these newer players.

Integrations and Trends

In this evolving automotive environment, new ideas are available for product-based achievements in seamlessly incorporating audio in seats, trim, and other boundaries in the car.

Sonified by Trèves is taking the integration of audio hardware further in a new direction and, with the support of a Dirac OPTEO tuning, it was demonstrated under the Audio Foundry’s tent in the Central Plaza at CES 2024. Trèves is an automotive supplier of interiors and acoustic treatment, specializing in recycled textiles for automotive.

Unlike the typical application of exciters to rigid door or dash panels, which over time can develop buzz, squeak, or rattle problems, Sonified selects a panel of the desired size and shape, attaching a transducer to it, and then suspending it in the foam underlayment of a carpet or a headrest. The air captured in the individual cells of the foam provide the equivalent of a larger space. That material is a polyurethane-covered honeycomb material like that used in package shelves.

The Sonified system was demonstrated in a Tesla Model Y that retained its complete stock audio system for comparison. The Sonified system used Tesla's three midrange and two tweeters, but turned off the door speakers in favor of two of these panel speakers placed behind carpet on the sides of the center console. The subwoofer in the cargo area was replaced by two subwoofers in the carpet under the front seats. And two Sonified speakers were mounted in each front headrest as well. These were able to reproduce sound from 60Hz to 12kHz. This also has an advantage in ANC or Personal Sound Zones.

Dirac OPTEO tuning will be paired with the all-new Trèves "Sonified" technology to showcase a future of automotive audio where smaller, lighter, and more strategically-positioned speaker systems can produce increasingly high-quality sound.

The Dirac OPTEO tuning features patented multiple-input/multiple-output (MIMO) mixed phase impulse response correction technology, addressing the acoustic challenges common in cars by enabling all speakers to work intelligently together and co-correct each other's impulse response.

The combination of Dirac OPTEO and Sonified's system design enables high-quality sound from a speaker system that's lighter weight and more strategically placed—which is increasingly critical for electric vehicles whose battery efficiency is largely dependent on vehicle weight.

Flexound Augmented Audio showcased an integrated seat sound solution at CES that would directly address the need for passenger personalization and immersiveness. After improving cinema experience in commercial and home theaters, the Finnish company took the technology and applied it to automotive seating, soon to be launching with a Korean OEM automaker. Their construction is two shakers integrated in the headrest, which also couples to the seating frame to generate a full audio band experience, which can also be tuned below 20Hz.

Flexound demonstrated the clarity of sound using multi-sensory listening. There was a clear improvement in speech and music perception and phone call quality. It provided a full frequency range with only one element, which can also produce stereo output. It too would enable ANC solutions with both sound and vibration.

Contributing to a new approach in hardware simplification, LG Electronics unveiled a transparent antenna for automobiles at CES 2024. The thin film broad-surface application is designed for compatibility with various types of glass and vehicle designs. R&D has begun using the thin film for reproducing information cues, not broad band audio, for the passengers and driver.

At its dedicated booth for automotive displays in the West Hall of the Las Vegas Convention Center (LVCC), LG Display showcased a series of ultra-large automotive display solutions, including the



Sonified By Trèves demo at CES 2024, tuned with Dirac OPTEO: exciters diffuse sound across composite surfaces decoupled from the hard surfaces of the car.

world's largest automotive display—the 57" Pillar-to-Pillar (P2P) P-OLED, and the largest slidable panel, the 32" Slidable OLED. These displays are an extension of LG's vision for each stage of the development of Software Defined Vehicles: Shift, Evolution, and Beyond. The crucial "screenification" of SDVs—which LG describes as "a big smartphone-on-wheels"—is seen as strategic to enable drivers and passengers to control a multitude of functions.

To help achieve perfect harmony between the display and the cabin, a new "Decorative Film" is attached to the screen to resemble interior materials used in vehicles, such as carbon or wood, when not in use. In addition, LG's proprietary Thin Actuator Sound Solution presented last year, and which is just 30% of the weight and 10% of the thickness of a conventional car speaker, can be installed in various parts inside the vehicle, such as the display, ceiling, and floor, to produce an immersive sound experience.

Virtual Product Development

The digital twin method for developing software and other components in a vehicle was previously mentioned. The idea of having a computer simulation of a design stand-in for a physical prototype and using that digital twin to refine, optimize, and test the design has been in the engineering world for more than a decade now. Its maturity has advanced considerably, and its adaptation is accelerating.

Today's vehicles contain far more sensors and embedded systems than ever before. The development of electric vehicles has brought radical changes to powertrain development, aerodynamics, and to the overall user interface. This in turn has shifted and



Flexound Augmented Audio complements existing automotive sounds systems.



The LG Display 57" Pillar-to-Pillar P-OLED, the world's largest single panel for automotive displays, covers the entire dashboard from the driver's seat to the front passenger's seat. The dashboard display is complemented with a 34" P-OLED center screen and a 48" P2P LTPS LCD that allows drivers and passengers to see useful driving information while enjoying entertainment features.

elevated the emphasis onto certain disciplines such as noise, vibration, and harshness (NVH), to address new factors influencing noise and ride comfort. Disciplines such as NVH are critical to ensuring a quiet, comfortable ride in the absence of internal combustion engines and the associated traditional NVH challenges those present. It's not one challenge, but multiple challenges: New vehicle architectures, increasing complexity, new disciplines emerging, software-driven development, some disciplines are becoming more relevant.

Increasingly, engineers rely on software to help not only design the system components and vehicles with a new complex architecture, but also to design the new vehicles better, faster, and with less cost. A major barrier and hurdle to accelerating vehicle development—and indeed vehicle innovation—remain in the bottleneck of physical testing. Testing physical vehicle and component prototypes has been central to vehicle development since the beginning of the car industry. However, with the mainstream adoption and rapid evolution of sophisticated software and simulation tools, the need for physical prototype testing is diminishing. There are significant advantages to reducing physical testing that extend beyond just the time and cost elements, especially later in the development process. Employing simulation tools during the development process allows for easier design iterations and lower cost than relying only on physical prototypes. The benefit of virtual prototyping is the role it plays in accelerating the pace of innovation. The ability to make late changes, iterate on designs that can be virtually tested, and the ability to incorporate human feedback—all of this enhances the designer's ability to innovate quickly and efficiently.

However, before these sophisticated tools can be effectively employed and the impressive gains in time, efficiency, and cost be appreciated, it is critical that the manufacturer first completely understands its own development process and where the company is in the "maturity level" of its development process. It is best for those that offer or supply the tools for digital twins to walk through

their own specific Virtual Development Process, showing their own maturity level and then provide a manufacturer or a customer with a map outlining their current development process, highlighting where they are in terms of using advanced simulation tools.

A more mature external process is a way to accelerate a manufacturer's internal development, linked to that more mature Virtual Development Process. After all parties understand where they are in the maturity landscape, a combination of dedicated training sessions can be provided, along with step-by-step mentorship in the preparation of design scenarios and data sets. This is an ideal way to be seamlessly introduced to the world of simulation in an organization where little or none exists. This also ensures the necessary skills to create and validate models, be it for an ADAS unit, HMI, algorithm, sensor, microphone, or loudspeaker. This applies to all product development inside automotive, the car and the audio system, and in consumer and pro audio.

Another challenge to achieving a large-scale virtual process development is the breaking down of silos that remain internally at organizations where each department has their own "digital twin." In automotive, there can be NVH twins, Antenna twins, Audio twins, and so forth. After those silos are broken, there's a need for coordinating with suppliers for data and model exchanges. What might be seen as the uncomfortable nature of, say, a traditional loudspeaker design engineer, who becomes a "subject matter expert" instructing and reviewing remote engineers doing the design work near a factory somewhere else in the world, is a step toward making digital twins and data science more accessible so that a "subject matter expert" is able to become the champion of a new idea and a collaborative design.

What's Next

In the coming year, we should be seeing new research and developments in directional perception of virtual sources, perception based on acoustical parameters, the influence of masking noise on spatial perception, and perceived spatial extent of local active noise control for broadband disturbances. We should experience the virtualization of subjective evaluations, virtualization of sensing strategies for active control of road noise, advancements in automated spatial tuning and virtualization, as well as enhancements to the height experience in immersive audio. And crucially, we should see innovations in sound design for automotive engine sound. 

About the Author

Roger Shively is the owner of Shively Acoustics International in Seattle, WA. He has more than 38 years of experience in engineering research and development, with significant experience in product realization and in launching new products at OEM manufacturers around the world. Roger co-founded JJR Acoustics in 2011 and had been a principal since its founding. Before JJR Acoustics, Roger worked as Chief Engineer of Acoustic Systems as well as functional manager for the North American and Asian engineering product development teams in the Automotive Division of Harman International Industries Inc., a journey that began in 1986.





DSP
Concepts

Conferencing Solutions

Available for the full range of enterprise and home conferencing products

- Edge of the room
- Center of table
- Ceiling mount
- TV
- Set-Top box
- Gaming

Reference design and ODM partners

AmTRAN

TYMPHANY

SoC vendors

AIROHA

Qualcomm

NXP

- AI base advanced conferencing solution
 - Noise reduction
 - Dereverberation
- Built-in and Remote mic support
- Direction of Arrival to identify a talker
- Virtual Co-viewing with far-field full-duplex communication
- Dante, AVB, A2B or USB interface
- Pre-integrated with leading SoC and DSPs
- Pre certifications support for Teams V5, Zoom and ACM



Automotive Solutions

DSP Concepts leverages decades of automotive audio signal processing design experience as the leading enabler of software-defined vehicles, providing transparency and control over every facet of the automotive audio experience.

- Multicore supporting concurrent usecase enablement for all audio uses on one device
- Support for Qualcomm and Exynos SDKs



Contact us to meet at AES Automotive Conference or Infocomm



Integrated Audio Development Frameworks

By
Philipp Paul Klose

Developers and designers are familiar with a number of audio frameworks that enable more time for actual innovation and less time to write code, debug, and test hardware deployment. New and existing integrated development environments (IDE) are essential as automotive audio evolves in tandem with software-defined vehicles.

The interior of the Audi Q6 e-tron.
(Image Source: Audi)

The software defined car is currently one of the biggest transforming factors for the automotive industry. The race is on to deliver a myriad of features on flexible, always-connected hardware, supporting over-the-air updates (OTA) and a flexible electronics system architecture that is protected from vendor lock-ins and allows for swifter upgrade cycles. Audio is on the forefront of change due to its key role in enabling a multitude of features in vehicles. To enable the change, a solid software foundation for audio applications must be used: the automotive audio framework.

What Is an Audio Framework?

If you have ever worked on a circuit for audio signals such as the DIY examples published in this magazine, you have probably discovered that components are soldered onto PCBs relatively quickly. In fact, for the completion of the project, a significant part of the work is not in the audio processing, but making sure you are building a safe and usable product. That involves devoting time to tasks such as casing, wiring, the power supply, and faceplates with control interfaces.

The same discrepancy in overhead-to-effect ratio can be found in software products. This is the case in software-driven products for digital mixing boards or in pure software products such as a DAW plug-in. Designing a proper underlying software architecture is a challenging job, not because of the features but because of the invisible software

needed to make it executable on the target platform and performing tests during the development stage.

Most basic, but fundamental audio algorithms—filters, dynamic processing, and transforms of different types—require between one and 30 lines of code for the pure processing part. The infrastructure to make these components usable is much more voluminous: pushing samples in and out, handling sample rates, device drivers, storage, graphical user interfaces (GUIs), visualizations, and other numerous tasks that require many more additional lines of code.

When the product is ready to be tested on different platforms with real-time processing enabled, profiling, test drivers, and an automated test suite fitted, the workload for the developer increases significantly.

It is at this point that audio frameworks become relevant: The concept of an audio framework is to provide developers with a selection of pre-written code snippets in an easily accessible library, allowing the ability to wire these modules together, and to make optimizations for processing hardware on DSPs, general purpose CPUs, and microcontrollers (**Figure 1**). The goal is that the user focuses on what really matters: the audio core of the product and the user experience. The essential groundwork for that to happen is performed on the background by the audio framework.

Special Frameworks for Automotive

Even if you have little interest in audio software engineering, you are probably aware of the sheer

amount of software solutions available for audio application development.

But why does automotive need special audio frameworks? The answer lies within the requirements of automotive engineering. The automotive world is highly regulated and conditioned by laws from all around the world. For every auto part there is very likely a law regulating the design in one or the other way. And the same goes for audio and software. Audio systems are nowadays driving the homologation of horns, acoustic vehicle alerting systems (AVAS), and other relevant systems. These must be super stable and able to boot up very quickly to fulfill the requirements determined by regulators. The hardware design is always of an embedded type, so the code itself must be optimized and compiled for microcontrollers and specialized CPUs and system-on-chips (SoCs), which often come with their own quirks and features.

One of the many factors making the software-driven car so compelling is the fact that signals can be sent anywhere. Theoretically, the engine control unit could know what the status of the rear-left window is and at the same time have the status of the indicators checked. That enables new applications, which are then rolled out to the driving fleet via OTA updates. No visit to the workshop is necessary to install a software update to exchange the sounds played from the instrument cluster, as an example.

Nowadays, audio is an enabler for many of the car's features: entertainment sound, phone services, and speech assistants are standard. In the future, many new functions will use the audio system for specific purposes. Just as a glimpse of what's coming, we have advanced driver assistant systems (ADAS), advanced noise cancellation up to midrange frequencies, auditory displays (what is happening in the driving situation and what the car is about to do), sound zones, and much more. All these systems must access the audio system, have priorities determined, and must be reliably managed. A tricky part for audio system engineers is the need for a number of sensors (accelerometers and microphones) and actuators (speakers inside and outside). All these parts are laid out around the cabin, increasing the complexity of the audio system with several subsystems.

Making sure that everything runs smoothly requires a high level of effort. In fact, the complexity of the audio system is comparable to the light- and propulsion-system of the car.

When to Use an Audio Framework?

Is there a tech stack already provided and maintained by a third party that we can use? Or

should we make our own? This is one of the first decisions in an early stage of software development.

There are good reasons to follow any of the paths. Using an external tech stack accelerates product development and reduces specific workloads, but can easily add headaches because of deprecated releases, future extensibility, irrelevant features, licensing fees, or lack of support. Using a self-developed stack makes initial developments longer, but the product has a tighter fit since it is developed around explicit or implicit well-known in-house requirements.

Both these approaches often lack proper technical documentation, making future extensibility and maintenance harder.

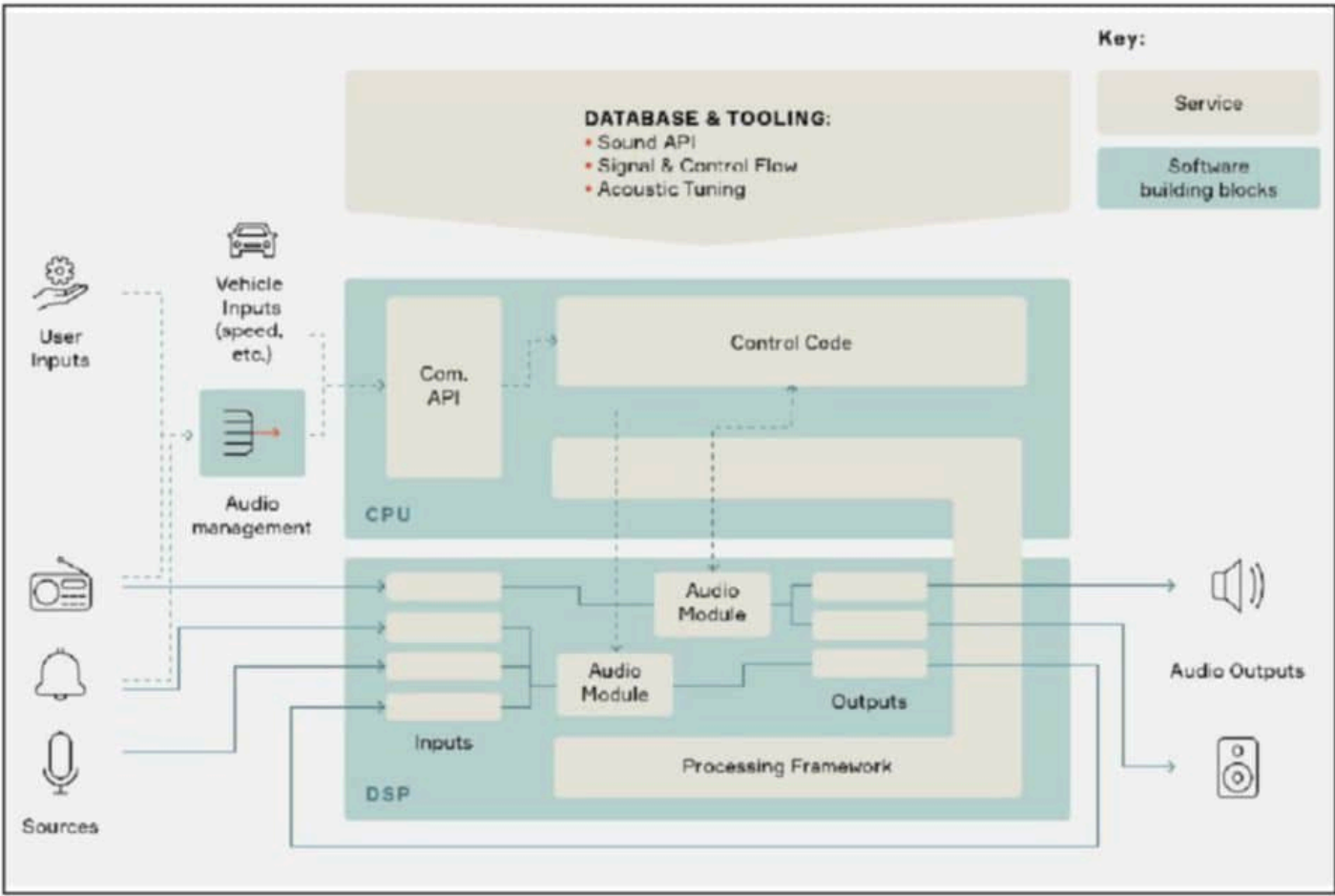


Figure 1: Technical architecture of a modern automotive audio system design by APTIV. The parts in green are the ones that are provided by the audio framework. The framework is using CPU and DSP as a resource. The DSP code is highly optimized to enable use of low-cost DSPs.



Figure 2: Analog Devices SigmaStudio+. This is the successor to the well-known SigmaStudio development environment.

Signal flows of audio systems can be very complex, even in relatively simple products. To make spaghetti code a thing of the past and the project easy to debug for the project team, all audio frameworks offer some kind of visual programming mode. Here the modules are “wired together” visually, greatly improving the creation experience. The tuning of parameters (e.g., filter settings, switches, and parameters for FIRs and biquads) can be manipulated in real time.

An audio framework makes sense when the company’s or the team’s core development activities are not in software engineering.

So, where to start? It is best to start with a list of known requirements and restrictions: What do we need? How fast? Do we want to implement code on our own? What hardware interfaces do we have to work with and what is the target hardware?

When Not to Use an Audio Framework?

The sheer quantity of solutions available today makes it hard to justify the development of our own audio framework. But there are some corner cases that justify a decision for self-development. One of these cases is met when our product sells in such small quantities or scales so massively, that the licensing fees per unit for the framework outweigh the cost for creating our own targeted solution.

In some rare cases we might want to create features or use optimizations that are not easily done with an audio framework. Or we must interface our DSP with an exotic type of interface for a very special application. If we have those kinds of restrictions, we are probably aware of the pros and cons before the project starts.

If the decision is to explore using an existing audio framework, it helps to understand some of available options.

SigmaStudio+

SigmaStudio is one of the oldest and most prominent audio frameworks. It caters to professionals and enthusiasts alike and is well established in many parts in the audio industry, including automotive. Many automotive projects use DSP software created with Analog Devices’ framework. Recently, Analog Devices decided to cut old ties and to release the overhauled version SigmaStudio+ (**Figure 2**). The editor software not only allows the creation of signal flows but can also be used to design A2B and future A2B 2.0 networks. The fresh user interface of SigmaStudio+ has a low entrance hurdle and is a good start to familiarize yourself with an audio framework. Readily available evaluation boards may act as step-ins when a product hardware target is not available. Being a tool from Analog it is obviously geared toward ICs from Analog and is solely used for future released DSPs and power IC programming. Through the CrossCore Embedded Extension, a wide variety of Blackfin, SHARC DSPs, and ARM designs are supported as targets for deployment. If you are not satisfied with the provided algorithms from Analog, you can add your own code through AddIns. Automated testing of logics is now possible through Apache Thrift support.

DSP Concepts’ Audio Weaver

Audio Weaver (AWE) is probably one of the best-known audio frameworks, created by DSP Concepts. When it comes to available functions, library size, and the number of board support packages, it is hard to beat. The AWE designer runs on top of a MATLAB runtime (Standard Edition) or a MATLAB install including the Signal Processing Toolbox (Pro Edition). Although MATLAB is available for

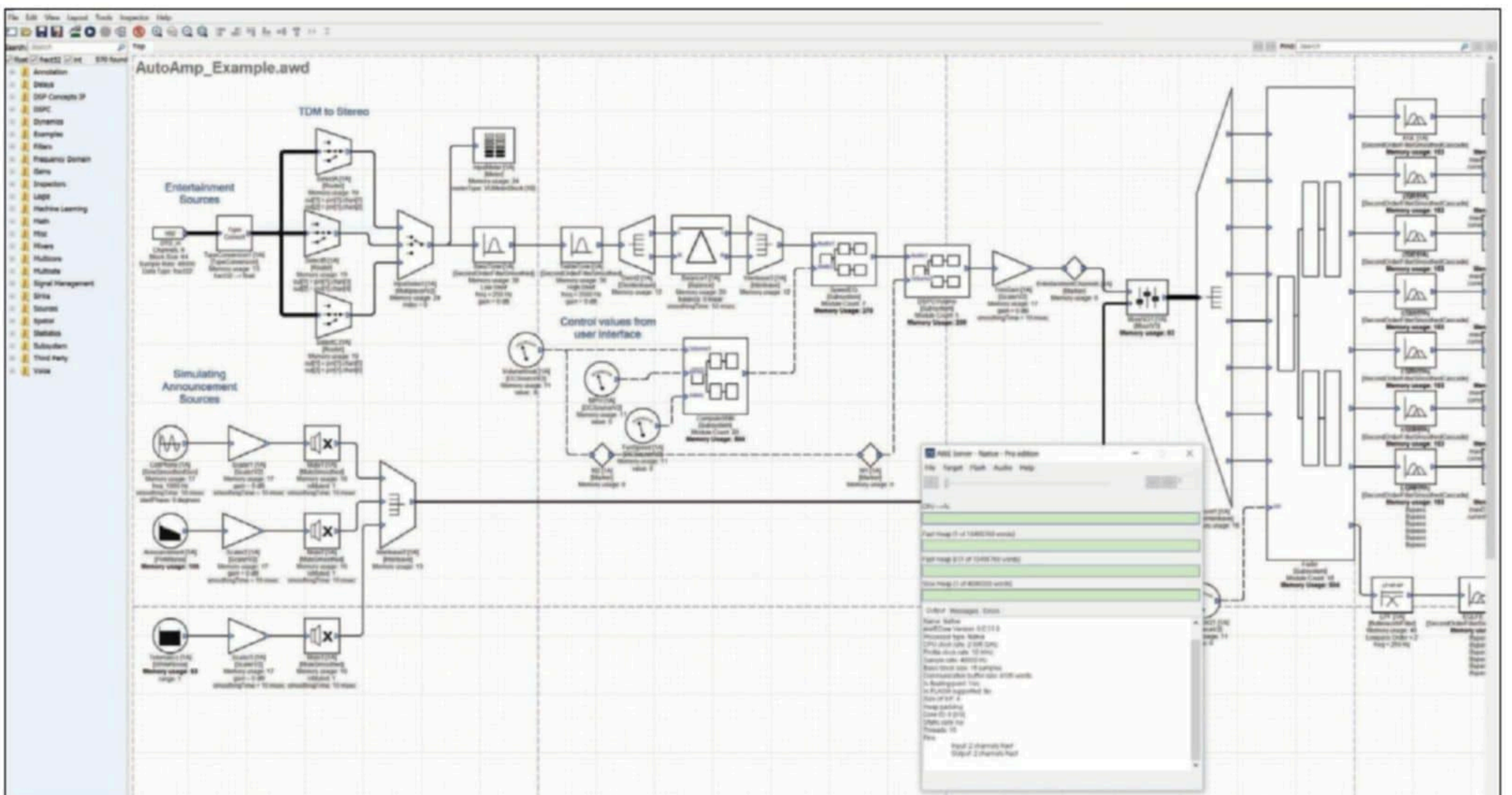


Figure 3: DSP Concepts’ Audio Weaver v8. This is one of the most powerful (automotive) audio frameworks available. The high number of modules and supported SoCs and DSPs make it a good choice in an environment where flexibility and speed is required.

Mac OS, AWE is not. MATLAB scripts can be used to test projects automatically and the audio functions from MATLAB integrate into AWE, making it a no-brainer choice for anyone who is proficient with MATLAB and its audio capabilities (**Figure 3**).

The code generated by AWE is highly optimized and uses very little resources. The AWE core application, which runs on the target, acts as an interface between the designer software on the PC and the signal processing application on the target. Many third parties offer their IP as modules for AWE.

QNX Acoustic Management Platform

The QNX Acoustics Management Platform (AMP) is a complete audio framework for Linux and Android-based operating systems (**Figure 4**). AMP is recognized for providing high guard rails in audio development. Many functions are preconfigured and straightforward for automotive applications such as chime generation, AVAS synthesis and speaker management.

Safety relevant applications are not a challenge, and it is not necessary to chain too many function blocks together. Complex functions such as road noise cancellation are already laid out in a helpful way. All features are tuned and configured via LiveAMP, which provides a complete overview of microphones, real-time vehicle data streams, loudspeakers, and more. One highlight in this package is the possibility to run and test generated code in a cloud environment, enabling virtual development operations.

While freedom in design is not the fortitude of AMP, it delivers a clear and concise package that can be deployed fast with a good set of selected parameters.

Harman's AudioWorX

Harman is one of the big names in Tier Ones for automotive and automotive audio. Brands such as JBL, Harman Kardon, and Bang&Olufsen are bundled under the Harman umbrella. Leveraging many years of experience, they came up with a design system that incorporates many components needed to create modern automotive systems (**Figure 5**).

AudioWorX takes a special role in this comparison since it is not freely available, nor distributed through a vendor. It is used internally at Harman and provided to selected partners. This is not a product you can download a demo and play a little with it. The

About the Author

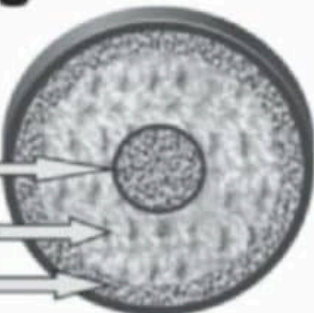
Philipp Paul Klose is a senior engineer for automotive audio concepts and functions at CARIAD SE and served before in positions as audio system development engineer, project lead and infotainment test engineer. He holds a Bachelor of Arts (Hons.) degree in Audio Production. Within his more than 10 years of automotive experience, he has worked with brands such as Audi, Volkswagen, Porsche, Bentley, Lamborghini, and Ferrari.



Spiral Airtm
Interconnects

*Coax
Is Cool
Again!*

Cardas Golden Strand
Air Filled Cotton Dielectric
Unique Spiral Litz Shield
ETI Tellurium Copper Connectors



Affordable Litz Interconnects and Bulk Wire
Tonearm - Phono - Line - Loudspeaker

KAB
Preserving The Sounds
Of A Lifetime
www.kabusa.com



flextech
— AKT



**Scalable A2B
test options**
from pocket to production

FlexTechAKT.com

Figure 4: Blackberry QNX is a framework with clear function implementations. Many use cases and requirements are covered through its Audio Management Platform.

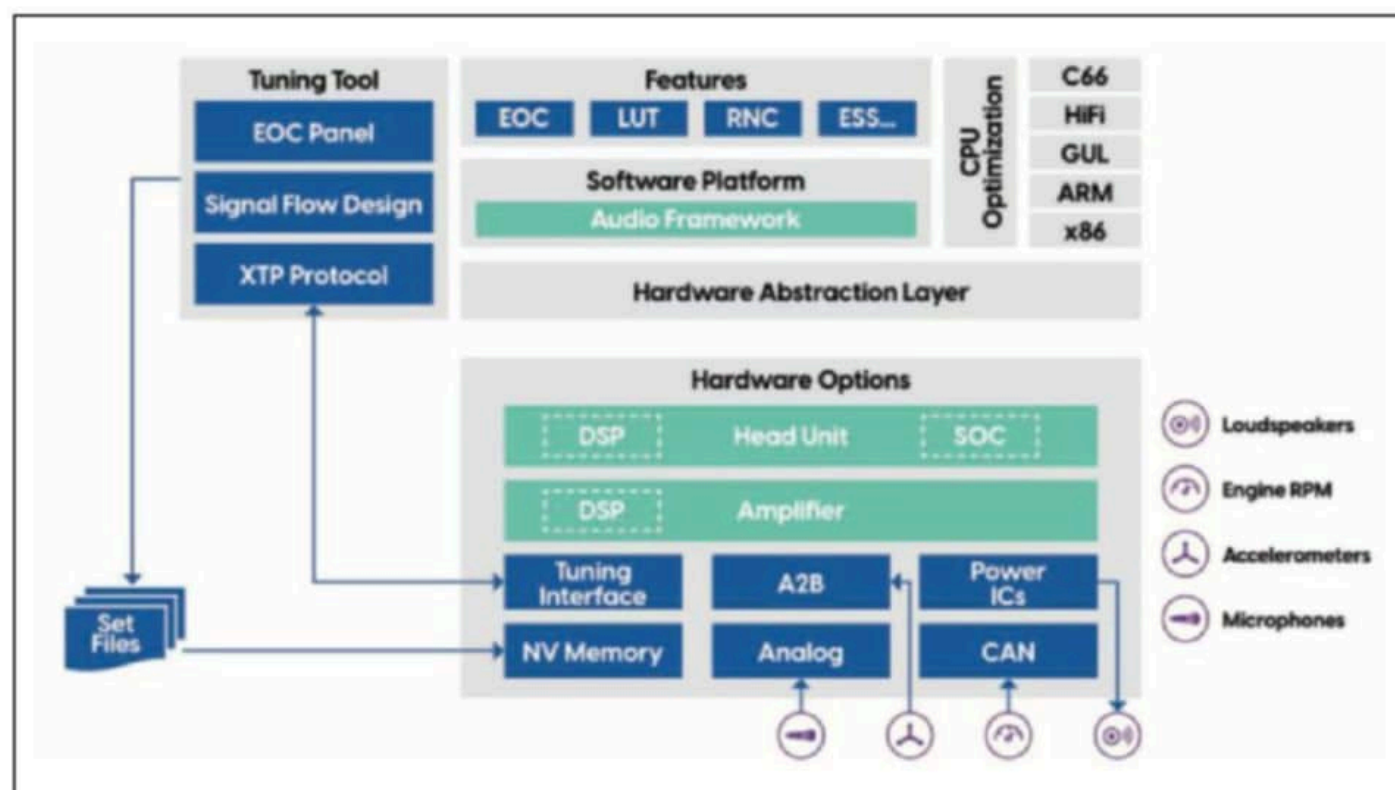


Figure 5: Harman AudioWorX system design. Optimizations for various DSPs from Texas Instruments, Analog Devices, Tensilica, and general-purpose processors from ARM and Intel are included. The framework powers all the sensors and actuators associated with the system.

license and costs must be negotiated. One of the key features is the integration of an acoustic simulation environment, through which the use of prototypes for testing and tuning can be reduced. Schematics can be test run on the target or natively on PC.

crispAudio's DroidSonic

Android Automotive gained massive popularity recently. Some automakers already have cars with this kind of underpinning in their product line-ups. Android apps can be executed natively here, bringing the user's digital life with known applications such as Google Maps or Spotify.

Android's audio system was originally not made with automotive applications in mind, which results in a relatively high latency, support for only one media source at a time and limitations in safety critical operations modes.

To solve these issues, crispAudio set out to develop an adapted audio stack and an audio framework, fulfilling the requirements of automotive. Through a hypervisor, the required processing blocks are split into several domains (**Figure 6**). The safety domain can withstand a crash of an hypervised OS and the low-latency domain enables road noise cancellation (RNC) and AVAS applications. Entertainment sound, radio, music streaming, third party algorithms and asynchronous sample rate conversions (ASRC) are all handled within the main audio domain, where the code generated via external frameworks such as Audio Weaver or SigmaStudio can also be nested.

Resources

Analog Devices' SigmaStudio+ | www.analog.com

APTIV | www.apativ.com

Blackberry QNX Acoustic Management Platform | <https://blackberry.qnx.com/en>

crispAudio | www.crispaudio.de

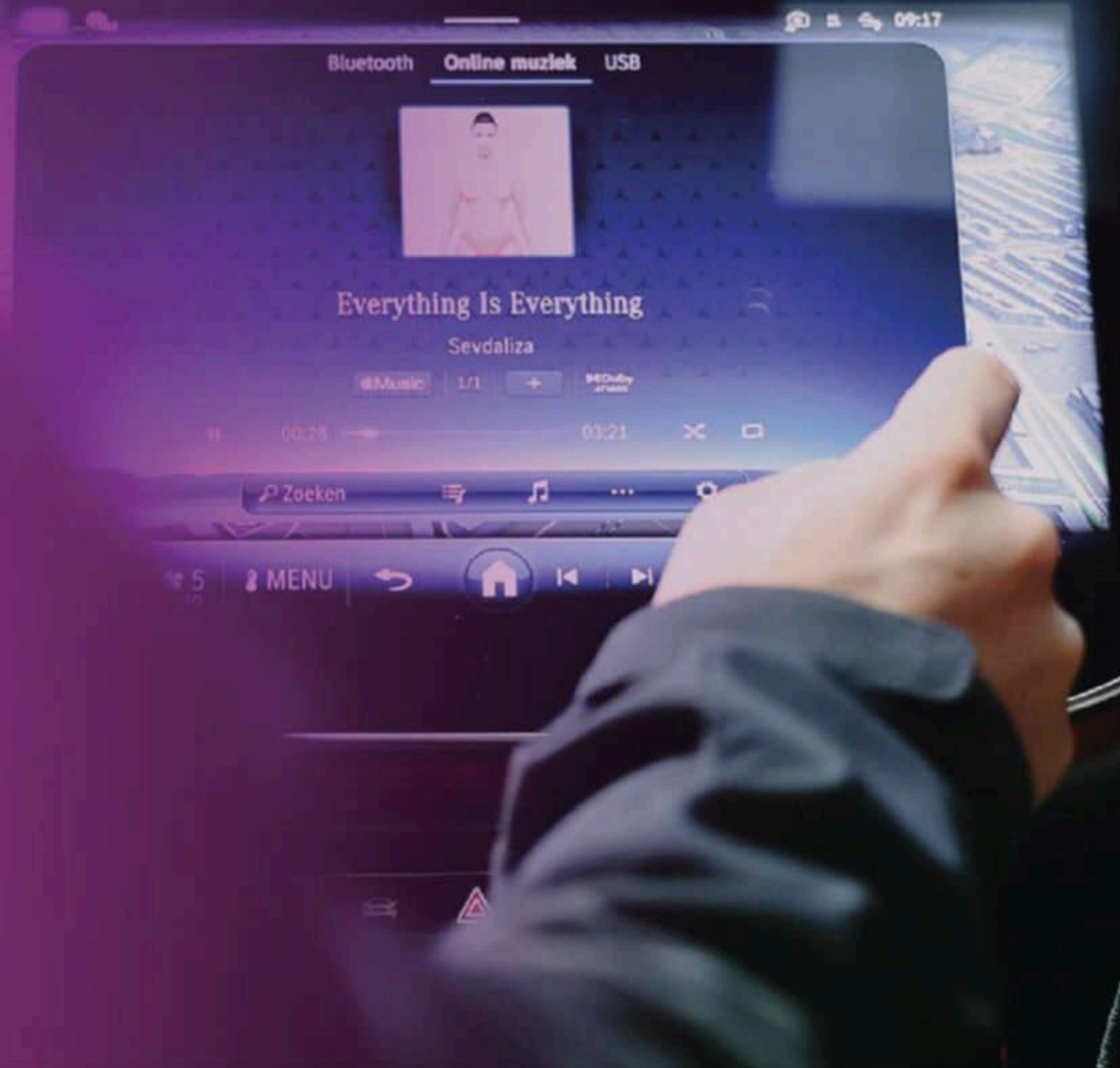
DSP Concepts' AudioWeaver | www.dspconcepts.com/audio-weaver

Harman AudioWorX | www.harman.com | <https://car.harman.com/solutions/car-audio/audioworx>

Impulse Audio Lab's EVx Suite | www.impulse-audio-lab.com



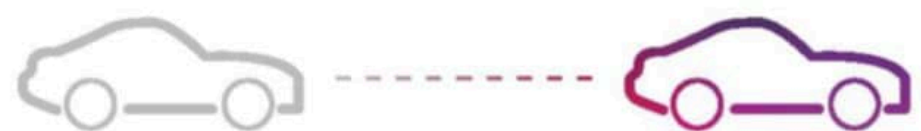
Your roadmap to the future of in-car entertainment



Join the revolution in automotive audio

The Dolby Atmos for cars ecosystem is expanding rapidly with an ever-increasing number of car brands and models embracing **a spatial sound experience like no other.**

Audio technology specialists understand the importance of staying ahead of the curve. Dolby Atmos for cars is leading the way, not only delivers immersive music experiences but is **reshaping the entire in-car audio landscape** with podcasts and audiobooks. This creates new revenue streams and opportunities beyond traditional music playback.



From luxury sedans to SUVs, more and more vehicles are joining the Dolby Atmos family, offering drivers and passengers alike **a truly incredible audio experience** on the road.

Don't get left behind! **Step into the future of in-car entertainment** with Dolby Atmos.

Adopt Dolby Atmos



Impulse Audio Lab's EVx Suite and EVx Drive

Impulse Audio Lab is a Munich-based company that excels in automotive audio and sound design and condensed its more than 10 years of experience into its EVx Suite, a framework created as its own internal design tool, which now is also available to the public (**Figure 7**). Coming from a more design-oriented background, EVx Suite is slightly more targeted toward the creatives in the automotive audio world. Functions for designing chimes, warning sounds, AVAS and haptic feedback are included, and the legal compliance of AVAS sounds can even be checked within the application, without the need for extra software. EVx can be integrated into Audio Weaver as a plugin, and it can also be used without other frameworks on ARM-based audio processors, SHARC, and Hexagon DSPs. The deployment in the car on SoCs is realized through EVx Drive. Having automotive system requirements and

cost targets in mind, the architecture of the software is designed to be as lightweight and modular as possible. A native mode allows for simulation and design on a PC without a car, cutting development time and cost drastically. Expression System is an included scripting language that allows complex pre-processing of parameters for engine sound emulation including drive pedal position, power boosts and torque.

In Closing

If you are tasked with developing audio software for a vehicle, it makes sense to use an audio framework. It saves time, and in most cases, it reduces resources and costs. Developers can concentrate more on the core of the product idea and the user experience. And the best part is that there are now plenty of IDE platforms available that match any workflow and needs.

Figure 6: crispAudio's DroidSonic. This audio framework brings several processing domains together, including all the algorithms that fulfill low latency requirements such as ANC/RNC/EOC or microphone pre-processing. It also supports code generated via external frameworks such as Audio Weaver or SigmaStudio.

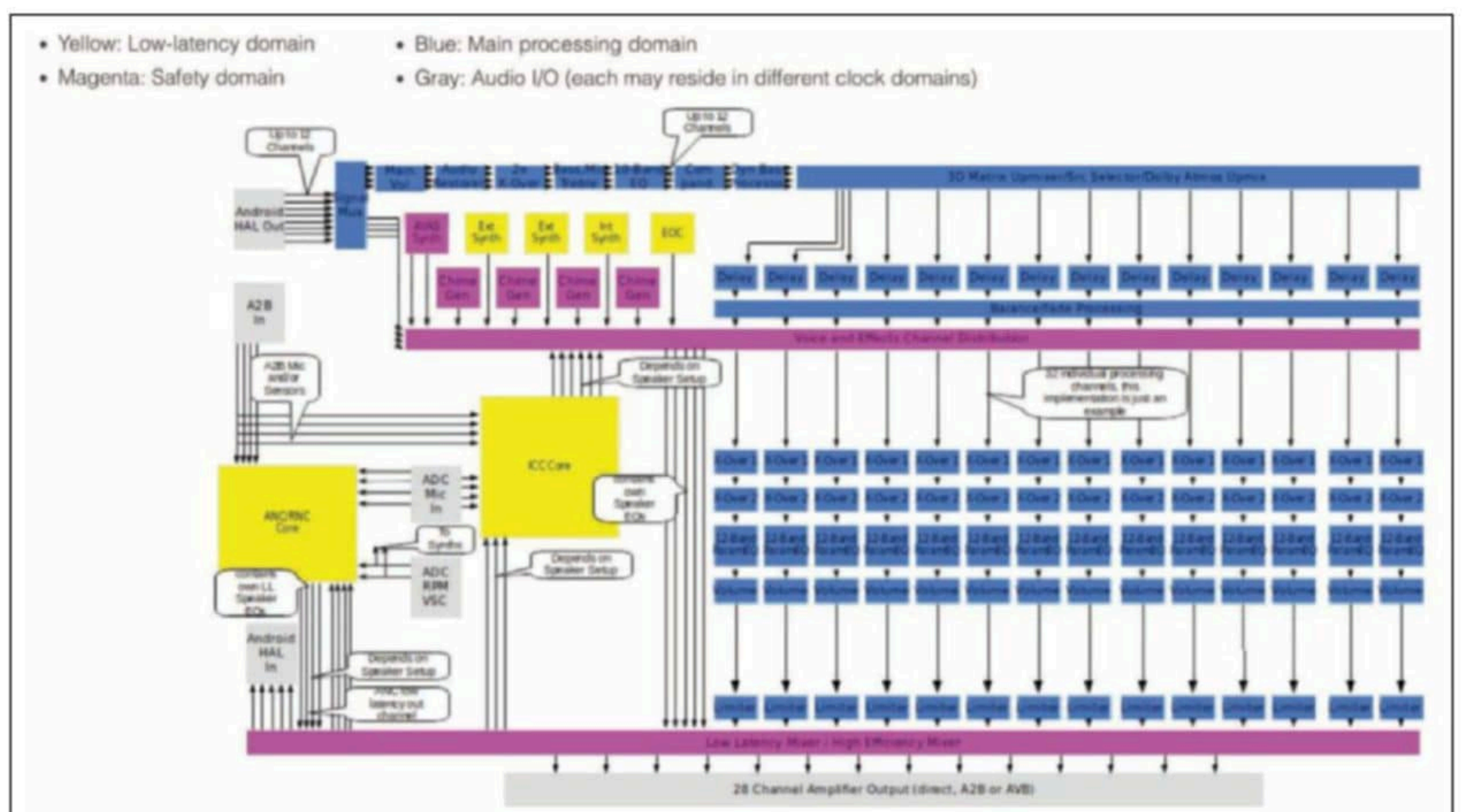


Figure 7: Impulse Audio Lab's EVx Suite. This interactive sound design suite has an intuitive graphical user interface (GUI), combined with optimized algorithms for digital audio signal processing in real-time.





***I can finally select
the info I need...***

using Bluetooth® LE Audio **Auracast™** – Reference-tested by **HEAD acoustics**



REAL-LIFE-PROOF.

www.head-acoustics.de





Harmonizing the Future

Crafting Immersive Driving Experiences for Electric Vehicles

By

Jasper de Kruiff, David Speith, and Lukas Joas

(Impulse Audio Lab, Icon Incar)

Photo 1: The Impulse Audio Lab development vehicle. Tuning driving sounds in real-life is essential to creating captivating sound experiences that match vehicle dynamics.

Vehicle sound design is essential to ensure that the transition from combustion engines to electric power doesn't compromise the thrilling essence of driving. For this purpose, Impulse Audio Lab created EVx, a specialized software suite for the creation of dynamic and complex sounds. Haptic seats developed by Icon Incar add synchronized tactile feedback and create a compelling synergy of audio and haptics.

Internal combustion engines have long been revered for their ability to engage the senses, offering a symphony of power that resonates deeply with enthusiasts. Whether it's the thunderous roar of a V8 or the high-pitched scream of a V10, the experience is more than just auditory—it's visceral, sending literal tingles down one's spine as piston banks pound away with relentless force. However, as the auto industry moves toward electric mobility, the distinctive roar is diminishing, posing a challenge to manufacturers in maintaining their sonic branding.

Due to this shift, innovative technologies are emerging to bridge this gap and redefine driving experiences for electric vehicles (EVs). This article examines the integration of interactive driving sounds with active haptic feedback, showcasing how these advancements revive and even surpass the level of excitement traditionally reserved for internal combustion engines.

Engaging Sound Experiences

Automotive sound design has evolved significantly throughout the years, originally enhancing internal combustion engines' sounds for improved characteristics and to establish vehicle

identities [1,2]. With the advent of electric vehicles, the challenge has shifted toward creating equally engaging e-sounds [3]. This can take many forms, from mimicking classic engine noises to futuristic soundscapes and even interactive music. Despite their variety, several key factors for EV sound have emerged that ensure a certain level of excitement and immersion for the driver (**Photo 1**).

These include dynamic responsiveness to driving actions, acoustic complexity to add interest and longevity to the sound, and nonlinear elements that encourage exploration [4]. The scope for sound design has also broadened significantly, extending well beyond engine sounds to a variety of sonification opportunities in and around the vehicle. This includes anything from exterior sounds when approaching the vehicle, engine start-up noises and even harmonized turn signals, all integral to crafting a coherent sonic identity. All these factors combined pose a significant challenge for automotive audio software, demanding solutions that strike a balance between creative freedom and efficient implementation to keep costs down.

To address these challenges, Impulse Audio Lab created EVx, a specialized software suite for

the creation, tuning, and integration of dynamic, complex sounds, benefiting from the team's combined expertise in interactive sound design, audio software development, and audio signal processing (**Photo 2**).

Crafting Sonic Experiences with EVx

Sound design for electric vehicles has seen a variety of approaches, including real-time synthesis, loop-based methods, and hybrids of the two. In real-time synthesis, live driving data, such as vehicle speed, is used to adjust synthesizer parameters (e.g., the frequency of an oscillator). Loop-based methods on the other hand involve the use of pre-recorded audio files. These files are then altered through gain fading, filtering, and pitching to match driving dynamics.

EVx relies on the loop-based approach to speed up the design process and keep resource usage to a minimum. Driving loops can be created from scratch with built-in synthesizers, such as the additive synth Genesys. It enables users to dial in a spectrum of harmonic overtones, which can be modified with a chain of different modulators and time-based effects for a wide range of possibilities from engine order replications to out-of-this-world sonic drones. An automatic looping algorithm takes care of rendering seamlessly looping audio files, ensuring a fast and iterative workflow. Alternatively, audio material

from any other source can be used, for example atmospheric soundscapes designed in a digital audio workstation (DAW) of choice or even the stems of a musical production.

The source material is then integrated into a driving sound tuning that features various interactive dependencies on signals coming from the vehicle, such as throttle position, speed, recuperation strength, or engine load. These signals can be accessed from the vehicle's Control Area Network (CAN) bus, a vehicle bus standard that allows the various electronic components inside the car to communicate with one another. The Drive section of EVx allows the user to assign both pitching and cascading of multiple loops to any of these CAN signals.

Hereby, pitching is realized by varying the playback speed of the loops and most commonly assigned to the speed of the vehicle. Cascading describes the way in which a volume crossfade is applied to a set of audio loops over a certain input parameter—typically engine load or throttle position. This enables dynamic spectral and timbral adjustments to the driving sound and forms the backbone of its interactive behavior.

High-quality driving sound tunings are characterized by multi-layered mixes that increase in intensity with more dynamic driving behavior. All these acoustic elements can be routed into one or

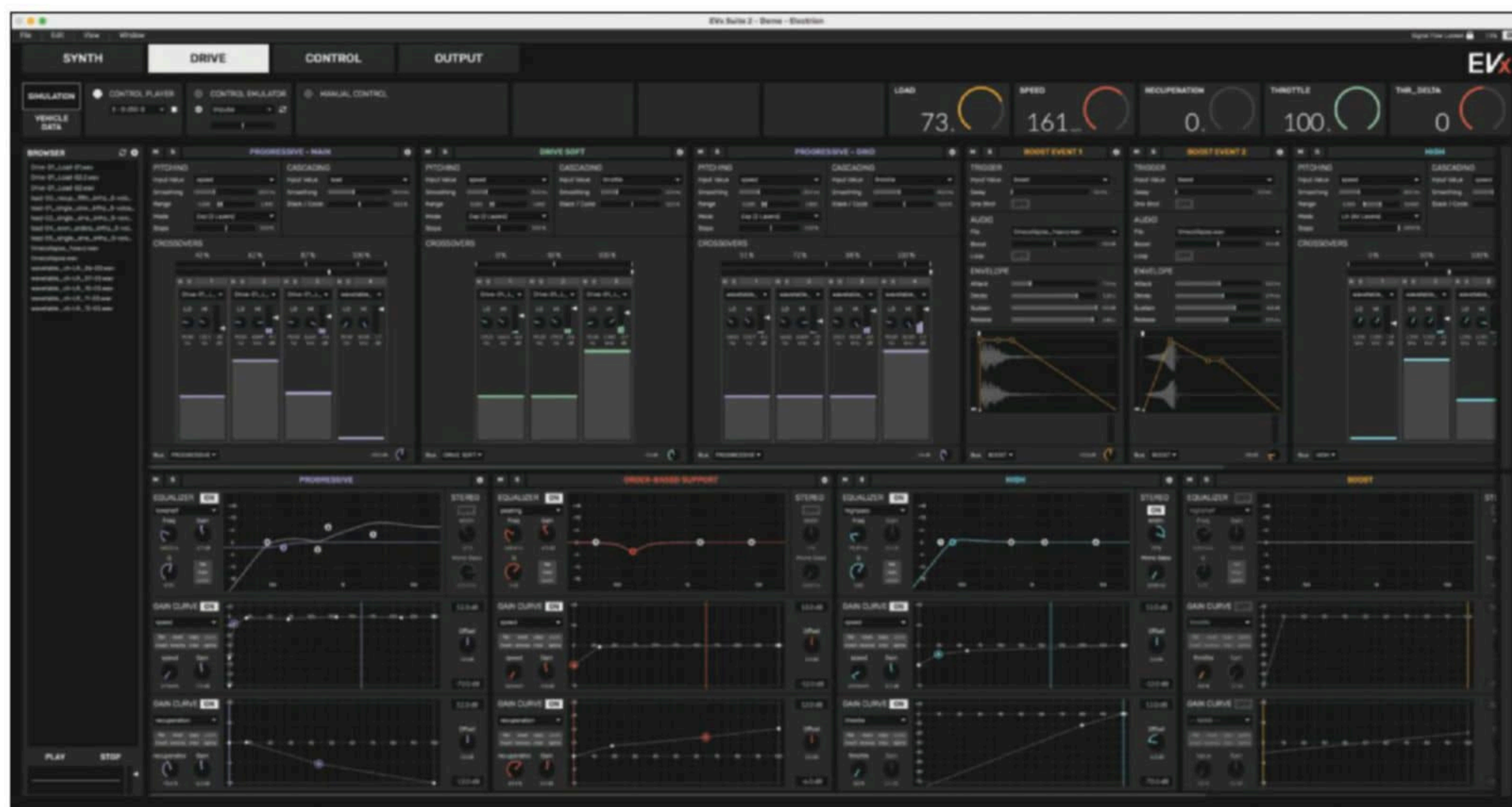


Photo 2: The Drive Section of EVx allows designers to rapidly create interactive driving sound tunings and transfer them seamlessly between desktop and vehicle.

more master buses, which allow for a fine degree of control using equalization, parameter control curves, and basic real-time audio effects. The output of the master buses can then be effectively routed to the individual speakers of the vehicle's audio system.

Shepard Pitching Behavior

Pitching loops by gradually varying their playback speed is an integral component of driving sound design, since drivers are used to hear rising pitches when accelerating and falling pitches when decelerating. However, a simple variation of playback speed is not enough to cover the entire speed range of an EV, which can well exceed 200km/h (125mph). One would either need a very high pitch range, which is not aesthetically pleasing, or a very flat pitch slope, which doesn't make sense semantically while driving.

In practice, Shepard pitching behavior is necessary to successfully implement the relationship between RPM and playback speed (**Figure 1**). This is an auditory illusion named after the musical phenomenon Shepard-Risset glissando [5,6], hereby multiple sonic elements rise in pitch exactly one octave apart, fading in and out in a manner that only leaves a continuous rise in pitch audible. EVx handles this sonically complex task fully automatically in the backend, letting the user adapt the Shepard behavior with high level parameters, such as pitch factors, slope, and maximum number of shadow instances. This approach eliminates the need for meticulously creating parameter curves by hand, speeding up the design process significantly and allowing for a more iterative design approach.

Enabling More Complexity in Sound

To elevate the driving preset to a holistic sound experience, designers can utilize EVx's event players for adding special effects and a full range of UX sounds. These players can trigger loops or one-shots based on specific conditions, which can be combined using logical operators, for example when triggering a launch control effect if both throttle and brake are fully pressed simultaneously.

For an even higher degree of dynamic control, the driving sound suite is augmented with an advanced CAN data preprocessing

system. This module utilizes a custom-designed C-like scripting language to perform fundamental mathematical operations, design complex triggers and envelopes, define custom functions, and much more (**Photo 3**). The resulting virtual CAN parameters can then be used as custom input parameters anywhere in the driving sound suite to create advanced sonic behavior, without the need for extensive DSP algorithms.

To stay in line with the requirements for agility and light CPU-usage, all the previously described modules and audio layers can be added or removed from the EVx session in a modular manner, making sure the software adapts to every use-case.

By enabling such complex behavior in EV sound, a solid foundation is established for the in-car driving experience. But to truly bring back the excitement of feeling your car's engine at work, we rely on the integration of a related but different feedback modality: haptics. To achieve this, we partnered up with Icon Incar, a leading mobility experience design company, to create Typ.1, an audio-haptic seat.

Enhancing the Driving Experience with Haptic Feedback

Icon Incar's Typ.1 proposes an augmentation of the sonic experience with haptic seats to enrich the driving experience with captivating body sensations. The feeling of tactile texture from the road beneath, of counter-steering the centrifugal pull from a curve or of being forced into the seat from acceleration are all part of an unspoken dialogue between driver and machine. This is a key element of what makes driving fun: There is a continuous exchange of control and response, with passive and active haptic sensations acting as a low-latency information channel.

What's even more intriguing: This channel serves as the only bidirectional interface to the brain. So-called muscle memory effects make intuitive interaction tangible and lay out a path to achieving mastery: A skilled driver will eventually not have to actively think about how to control the car given a situation, but rather draw from the experience anchored in the body and take control intuitively.

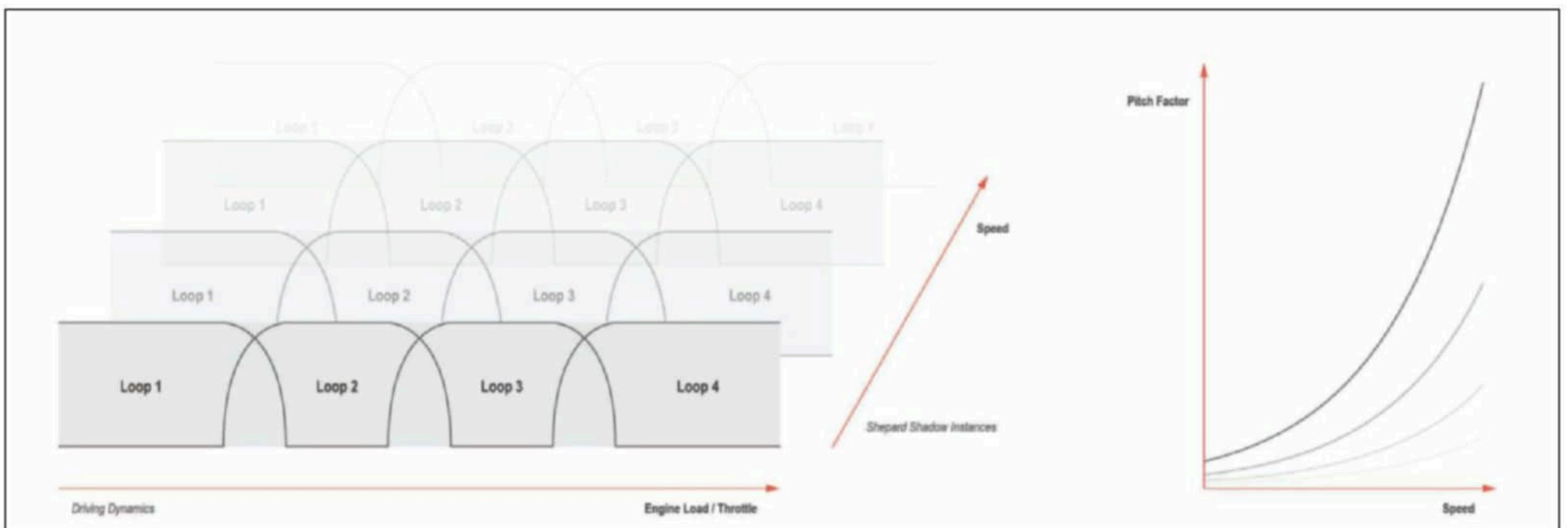
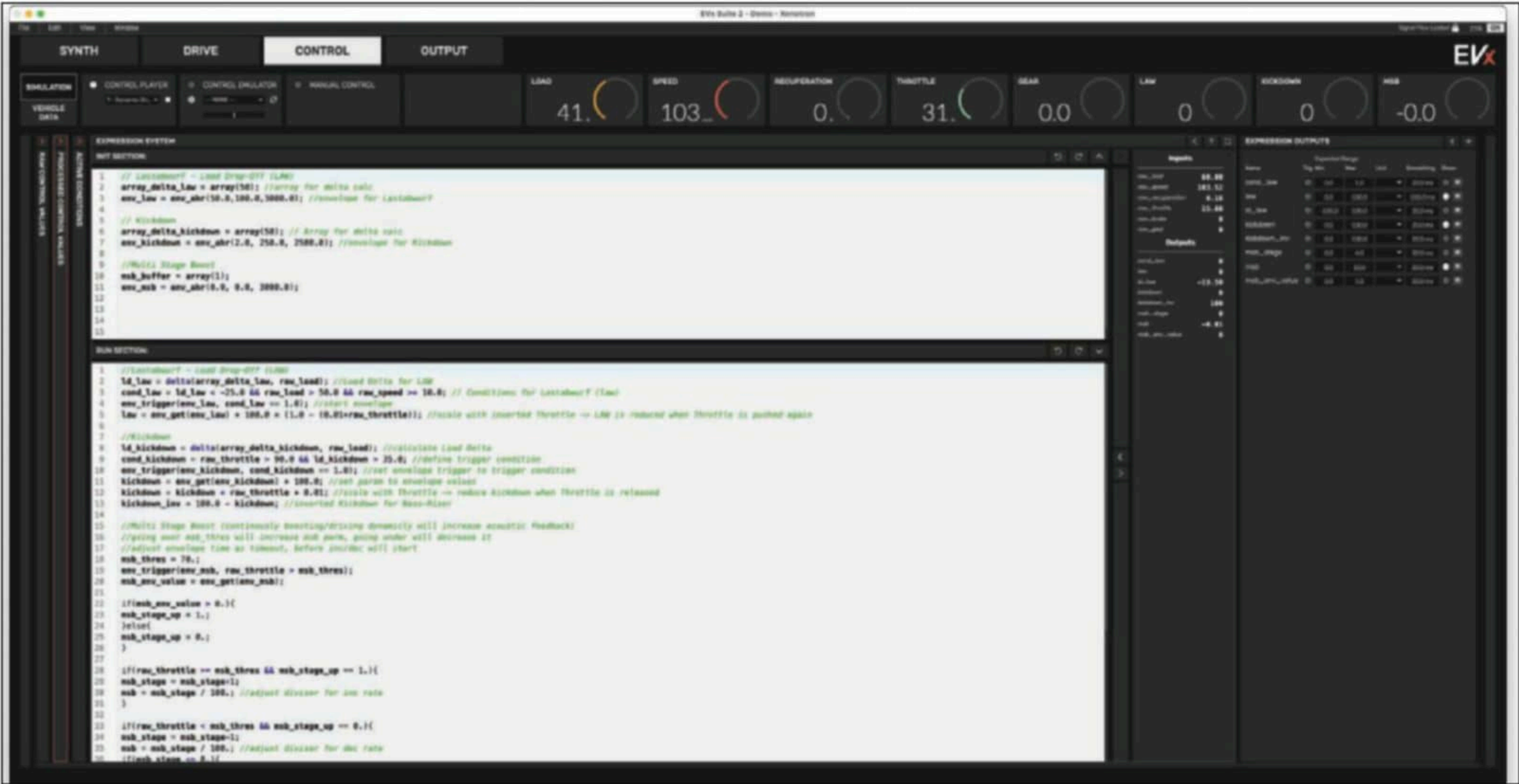


Figure 1: Shepard shadow instances create the illusion of an endless rise in pitch, which is essential to sonifying the entire velocity range of the vehicle.



Especially in electric vehicles, where traditional engine vibrations are absent, haptic feedback fills the void, offering compelling sensations: Imagine accelerating and feeling a subtle yet distinct vibration that synchronizes with the revving sound emitted from the speakers. This synchronized feedback not only enhances the immersion but also provides valuable tactile cues about the vehicle's state and dynamics. In a seamless integration with interactive drive sounds, haptics can unleash their full potential.

Through careful calibration and integration, haptic actuators embedded within the seats render dynamic signals, ensuring that drivers receive nuanced feedback that corresponds directly to their actions. By aligning haptic feedback with specific audio cues, such as acceleration or deceleration sounds, drivers can perceive a heightened sense of speed and power.

Likewise, a multitude of various dynamic aspects of driving the car can be nicely integrated: For example, in Typ.1 demonstrations, the braking sensation of an EV is accompanied by a subtle pleasant sensation, which communicates nicely how energy is being recuperated.

The synergy of auditory and tactile stimuli creates a holistic environment, captivating occupants and fostering a deeper connection within the vehicle. Various haptic designs can be tailored to support brand identity, communicate the drive mode (ECO vs. Sport) and accommodate specific customer preferences. This can include adaptive profiles tailored to individual driving styles and dynamic adjustments based on road conditions.

Sensory Augmentation Broadens Awareness

Haptic feedback can contribute to safety and comfort during the journey even more directly. By providing additional tactile alerts before and for critical events, such as lane departures or proximity warnings, haptic feedback enables drivers to stay informed yet to remain visually focused. Through the use of spatialized haptic feedback across strategically placed actuators, various information can be mapped directly to the driver's back. Initially supplementing visual and auditory cues, the human brain learns to process these tactile sensations more directly until they become integrated into the subconscious perception system over time. This phenomenon is called sensory augmentation and effectively extends the sensory capabilities beyond visual and auditory channels. This is how Advanced Driver Assistance Systems (ADAS) can be mapped more directly into the driver's perception thereby fostering a heightened and more intuitive awareness.

Photo 3: The Expression System is a C-like scripting language that allows real-time processing of CAN data with functions and mathematical operations to create custom control parameters.



Photo 4: A range of audio exciters and haptic actuator components.

Furthermore, as autonomous driving technology progresses, similar dynamic haptic feedback can play a vital role in maintaining a relationship between passenger and car. Our proposal also introduces a range of dynamic experiences: Particularly during long journeys, driver's and passenger's biometric data can be monitored, including their focus and stress levels. The system can then adjust the character and intensity of feedback from the seats accordingly. Sharper vibrations stimulate alertness if the driver is fatigued, while calming vibrations promote relaxation if stress levels are high. For these features, drivers as well as passengers will be able to personalize their haptic profiles for customized comfort and choose from a library of haptic sensations that suit them best.

Music Feels Better with Typ.1

Haptics also offer a unique opportunity for enhancing the entertainment and music experience beyond traditional audio systems. The captivating feeling of music at a concert being transmitted directly into the depths of the body can hardly be matched by the best audio systems of modern cars. However, this engaging musical effect can be approximated much better by Typ.1 haptic seats.

In contrast to classic subwoofers, haptics offer several advantages for music reproduction in car seats: Since actuators are placed directly at the desired impact zone the haptic effects can be tuned precisely to various parts of the body. Positioning

relevant vibrations in haptic and audible frequency range directly at the body is also more energy-efficient and therefore more cost-effective than transmitting bass sounds only through the air in traditional sound systems. Additionally, this technique does not drown the whole environment with bass and still allows for situational awareness and more comfortable conversation. The convergence of audio and haptic crafts a fusion experience that defies explanation and should be felt firsthand.

Engineering Haptic Seats: From Actuator to Perception

The placement, layout, and orientation of haptic actuators are critical for the overall potential of perceived experience. Thereby it is important to consider how the variability of specific actuator performance changes when integrated into the seats' cushions and foam. During the design process, the characteristics of actuators embedded into seats are measured. Once the environment is understood, specific digital signal processing is applied to the haptic signals of individual actuators to ensure the best possible performance. Human perception of different body parts needs to be considered and integrated in the signal processing chain depending on the body location too.

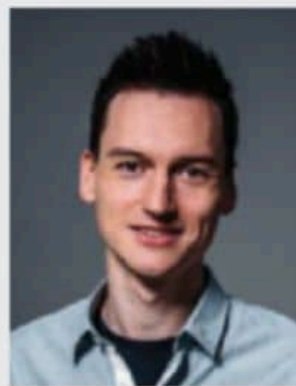
The live computing of signals and power amplification of actuators requires dedicated hardware, which can be easily integrated into the vehicle. The Typ.1 XPU - Haptic OS provides a solution that works with a very wide range of actuators and allows placement of low-frequency pure haptic focused transducers as well as more conventional exciters that reach into the audio range frequency range. Depending on the customer's focus various actuators from the use of simple single-frequency exciters, low-frequency speakers, and voice-coil actuators have been used to create convincing haptic experiences (**Photo 4**).

Drawing from our team's experience, it is vital to embrace a holistic approach to sound and haptic design to maximize the synergetic potential in future automotive experiences. Most natural sensations exhibit a tactile dimension, which is directly interleaved with its auditory counterpart. Since the goal is to create a similar natural engagement of the senses, aligned with the vehicle dynamics, direct alignment of sound and haptic character and intensity is key for creating the best, holistic experiences.

With the availability of vehicle-wide CAN data streams, it is a great practice to have both systems utilize the same control signals and react similarly in their computation of stimuli. Given their shared nature as oscillations, EVx's audio signal can also be

About the Authors

Jasper de Kruiff is creative director and co-founder of Impulse Audio Lab, a creative sound and engineering agency focused on audio innovation. He holds a Master Degree of Science in Industrial Design (Hons.) from the Eindhoven University of Technology and has spent the last decade combining his life-long passions for music and design crafting innovative audio experiences for electric vehicles.



David Speith is a seasoned embedded engineer who specializes in haptic interfaces and consumer products. His experience extends 10+ years in user-centered software development and achievements such as the creation of patented proprioceptive technology in the music tech industry. As an independent freelancer, he drives innovation in product development and interactive user experiences.



Lukas Joas is a sound designer at Impulse Audio Lab, working on interactive sound design projects for leading OEMs around the world. He has completed a master's degree in communication at the LMU Munich and has a background in audio engineering and music production.



fed directly into the haptic computation system for a more immediate linking of auditory cues and tactile sensations. Ultimately, the convergence of the best design hinges on the subjective experience of users. While scientific ideas can guide initial alignment, the true test lies in what the majority perceives. Thus, integrating user testing and rapid iteration loops becomes paramount, allowing for the real time refinement of audio and haptic experiences until a final design captures the most compelling experience.

What Is Next for In-Car Experiences

Thinking about future steps for in-car experiences, interactive music is often named as one of the most exciting trends. Being a cross-sensory experience, it is also suited as a playground for bridging audio and haptics, as music appeals to both auditory and tactile channels of perception. Haptic feedback plays a vital role here, compensating for the potential loss of driving cues when traditional vehicle sounds are replaced by music. It can communicate important information like speed or brake status through varying vibration intensities, thus maintaining driver awareness and safety.

As automotive technology continues to evolve, so too will the capabilities of multisensory feedback systems. By embracing the potential of sound and haptics, manufacturers can redefine the driving experience in electric vehicles, offering unparalleled levels of engagement, safety and comfort for drivers and passengers alike. 

References

[1] J. Scheuren, "Active Design of Automotive Engine Sound," *Semantic Scholar*, 2002.

[2] K. Weiinau, "Definition of the Sound Enhancement Preferences for a Sport Vehicle," INTER-NOISE and NOISE-CON Congress and Conference Proceedings, NoiseCon19, San Diego, CA, pp. 1-912, 2019

[3] J. Pruetz, "Active Sound Design Methodologies for Hybrid and Electric Vehicles," SAE International, 2021.

[4] J. de Kruiff, et al. "Dissecting Advanced Driving Sounds for EVs," presented at the Audio Engineering Society (AES) 2022 International Conference on Automotive Audio, 2023.

[5] R. Shepard, "Circularity in judgments of relative pitch," *The Journal of the Acoustical Society of America*, Volume 36, Issue 12, December 1964.

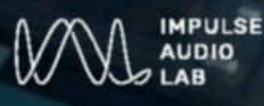
[6] J-C Risset, "Pitch Control and Pitch Paradoxes Demonstrated with Computer-Synthesized Sounds," *The Journal of the Acoustical Society of America*, Volume 46, Issue 1A Supplement, July 1969.

DRIVE THE FUTURE

Design, Tune & Integrate
Stunning Interactive
Sound Experiences for
Electrified Vehicles



WWW.EVX-SUITE.COM



www.sonical.ai

Listen to audio.
Listen to the world.
Listen to your body.

You decide.

Headphone 3.0

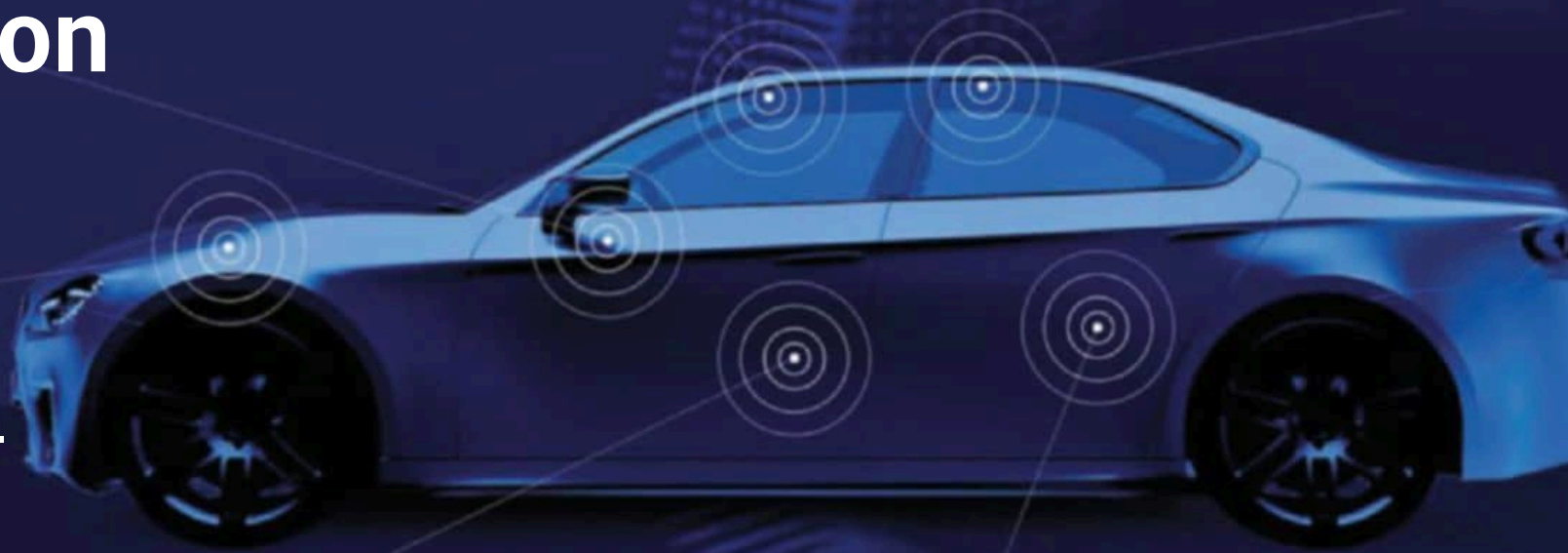


SONICAL
Beyond Sound

Car Audio Tuning and Upgrading

The Crucial Question

Aftermarket car audio always needs to be built on top of the audio systems that the manufacturers designed. This provides an opportunity to understand what works and what can be improved. A perspective that is always valuable for any designer.



By
Ken Ward

Many articles in this publication are written by acoustic engineers, employed by automotive audio suppliers serving the OEM automaker customers. My background is different—I have spent my career in the aftermarket installing higher-performance car-audio systems, and a great deal of my work in technical marketing for Elettromedia involves analyzing the optimization techniques used by OEM audio suppliers (this lets us take them into account during our own system-optimization work). For this reason, I have a rare opportunity to “see” various strategies used by OEM audio suppliers in the aggregate.

To oversimplify the difference between the aftermarket and the OEM system, the OEM system has severe constraints on current draw and weight, which combine to limit power amplifier performance (current availability, power-wire gauge, and heatsink size are all related to these variables). This limits the overall output of the system, as well as the low-frequency performance (additionally, weight limitations prohibit heavier subwoofer enclosures).

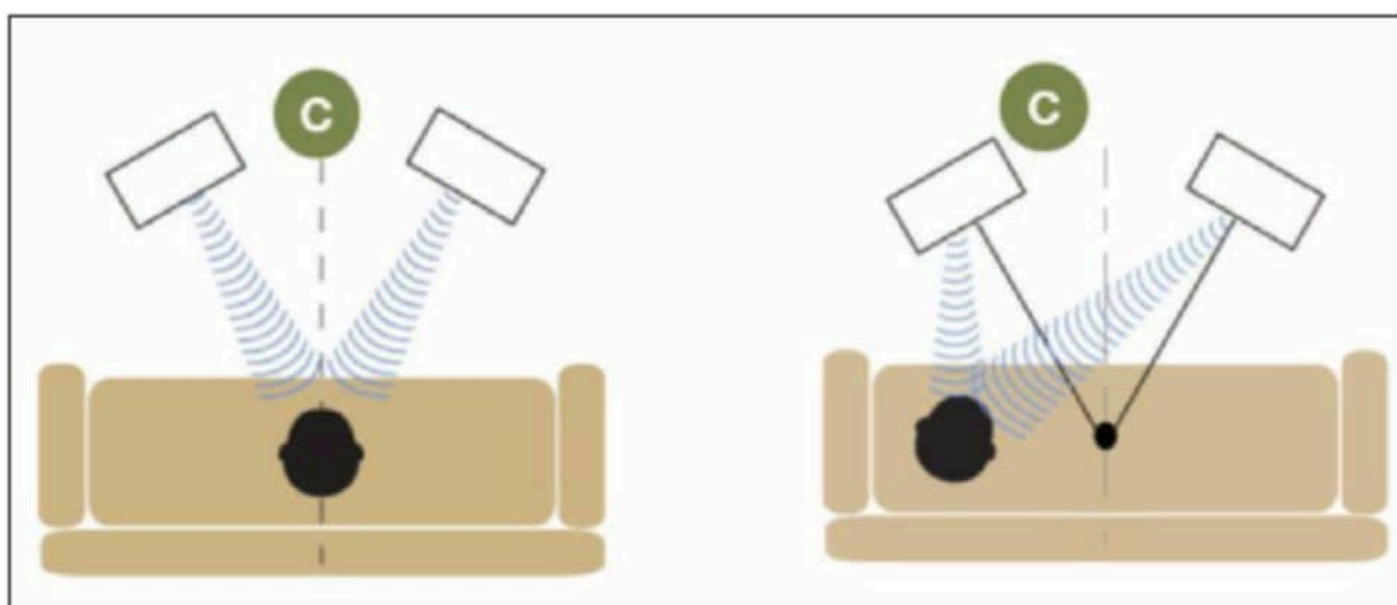


Figure 1: The first problem of car stereo—one side is louder.

On the other hand, aftermarket systems have generally been designed by non-engineers, and until recently have rarely addressed acoustic challenges in the car. The availability of affordable automotive-grade DSP chips have given the aftermarket tools, which we are still getting our arms around, so to speak. There is a great deal of education to be done.

In teaching automotive sound tuning to aftermarket technicians, I use a concept called “The Three Problems of Car Stereo.” All three of these problems are much more common in car stereo than in home audio and are caused by our predictably off-center seating positions in the car.

The Three Problems

Problem #1: One side is louder—Of course, that’s usually the near side, due to its proximity. This amplitude mismatch pulls the center image to the near side (**Figure 1**).

Problem #2: One side sounds different—Due to the off-center seating position, the left and right speakers may have very different angles relative to the driver’s listening position, and so the left and right speakers may have very different frequency-response characteristics. This causes the center image to float left and right, as the frequencies used by the performer go up and down in register, and the image floats toward the louder speaker at that frequency. The severity of this problem varies with the speaker system used.

For years, the most common front speaker system in vehicles was the 6.5” woofer/small tweeter two-way combination, but this system suffers from extremely compromised off-axis performance at the top end of the woofer’s range when the woofer is installed in the typical location (the bottom of

the front door). A driver complement, which works just fine in mini-monitor speaker cabinets aimed at the listener, loses its luster when installed in the bottom of a car door, nearly 90 degrees off-axis on the near side!

This upper-midrange dispersion weakness is exacerbated in many OEM systems, where the size of the tweeter is limited due to cost—thus limiting the low-frequency extension of the tweeter—and the resulting crossover point must be very high as a result. This unfortunately ensures that the 6.5" driver can't possibly meet the tweeter at the crossover point, especially on the near side (**Figure 2**).

The use of three-way, front-stage speaker systems in some premium sound packages handle this off-axis performance problem more directly—the 6.5" driver can be restricted to its omnidirectional range, and the superior off-axis performance of the small midrange takes care of these upper midrange notes. This does extract a cost penalty, since two more speakers are required, as well as two more amplifier channels to power them.

Another way to address Problem #2 is the now-widespread use by OEMs of a wideband midrange/treble driver paired with a 6"x9" woofer, with both wideband drivers in an off-axis orientation in the top of the instrument panel (as described by Roger Shively, et al, many years ago, and now found in many Toyota, Hyundai, GM, and Stellantis vehicles). Many systems in this category use off-axis wideband speakers facing up in the dash (for similar left/right levels and response), and door woofers operating in their omnidirectional range (for similar left-right response in the bass).

Problem #3: One side arrives first. The comb filter created by multiple arrivals of common-center information damages the frequency response for any content the recording engineer placed in the center of the stage. Since it is intended for the center of the stage, the content is contained in both the left and right channels, in phase, and at similar levels (**Figure 3**).

The Crucial Question

There are a few ways to address these problems, but first we must answer a crucial question: How many listening seats do we care about—one or more than one?

Solutions to the Three Problems

In how they address the Three Problems, OEM automotive audio systems fall into a few categories. Here we have presented five different types of stereo presentations for automobiles (**Figure 4**).

Type 1—Stereo (unmodified). This was all we saw back in the 1980s when I started, but it is increasingly rare today, since it doesn't solve any of the Three Problems. Stereo is of course symmetrical by definition, even if the Three Problems are significant in either seat.

Type 2—Stereo with a mono center speaker. Common in premium vehicles with entry- or mid-

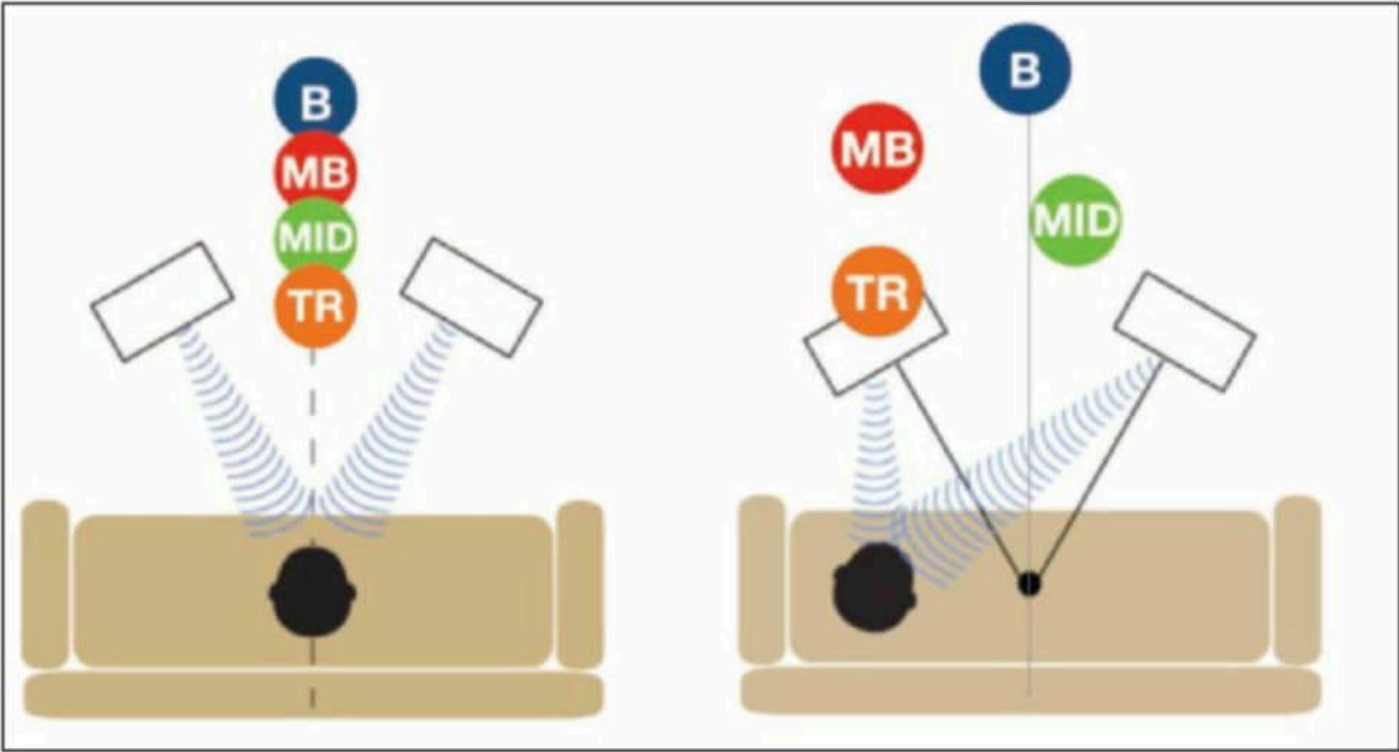


Figure 2: The second problem of car stereo—one side sounds different.

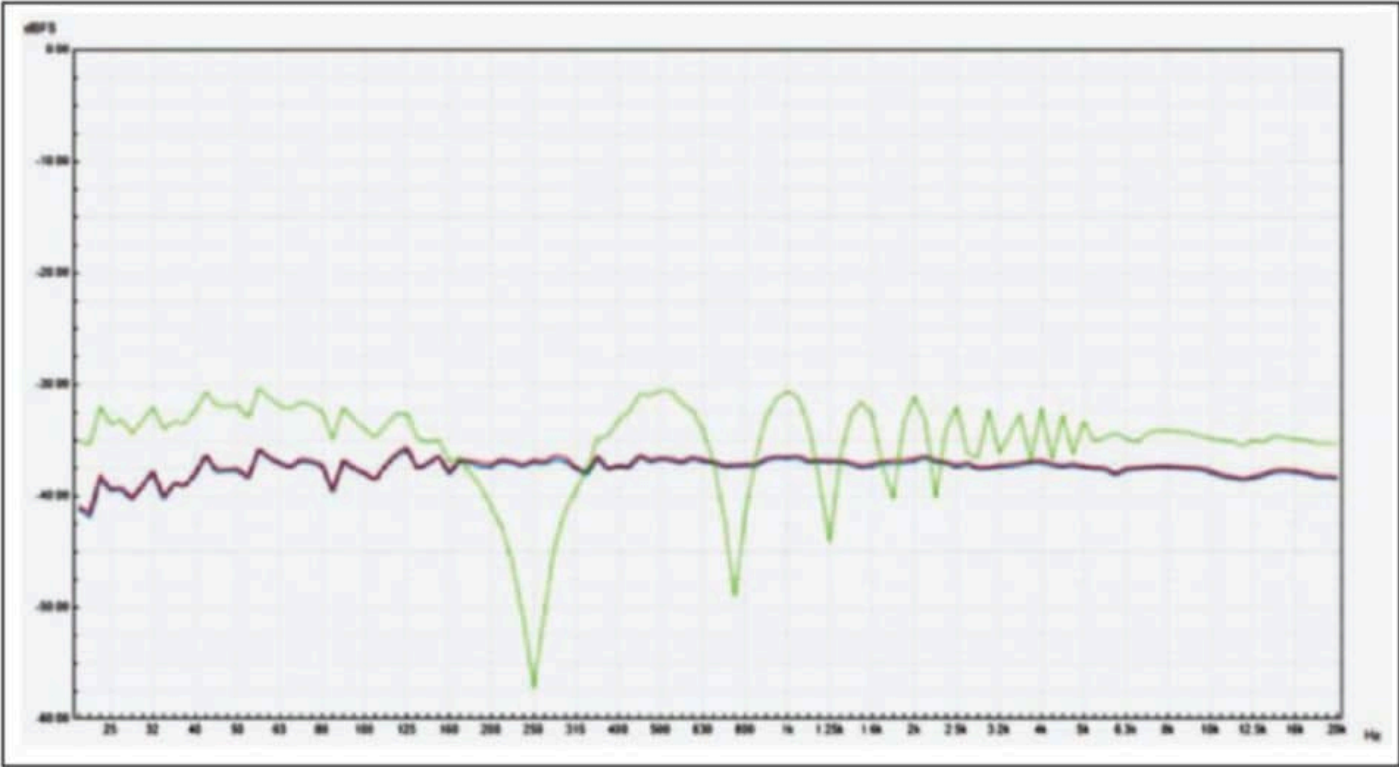


Figure 3: The third problem of car stereo—one side arrives first. This simulation shows the comb filter in the driver's seating position of a Toyota Tacoma (reflections eliminated).

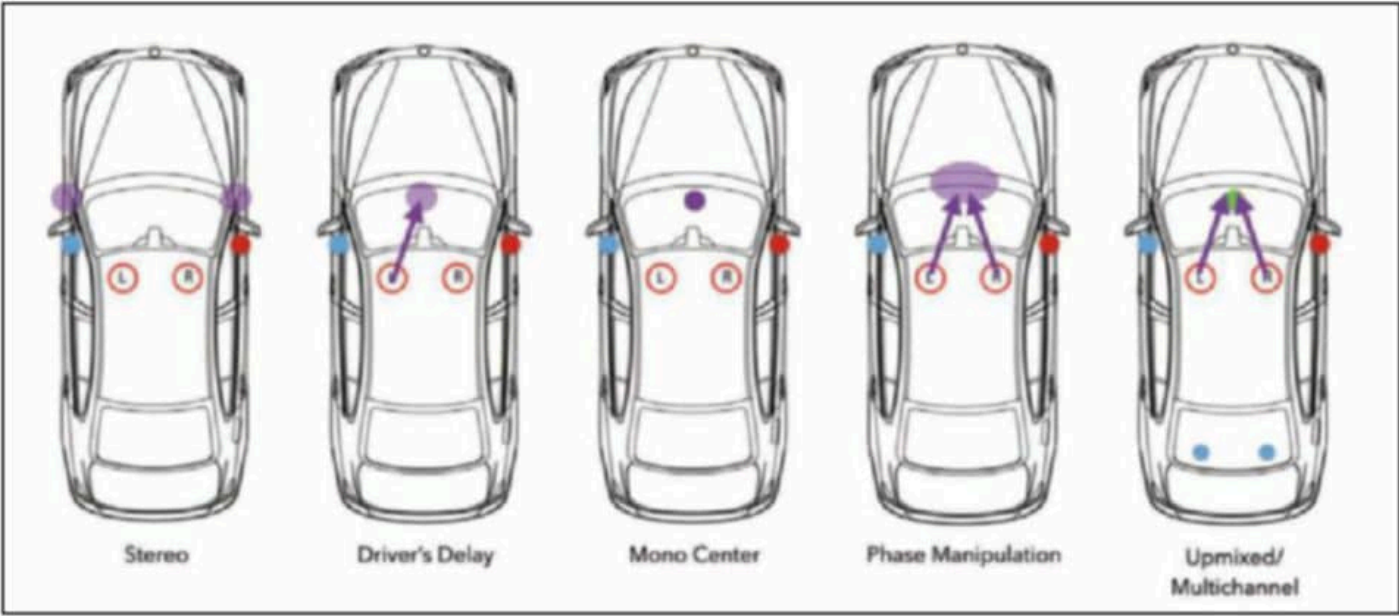


Figure 4: Five stereo presentations for automobiles.



level audio package, mostly just to fill the dash grille present for premium systems in Type 5 below. Not a good way to achieve great sound—the comb filter is still present and the stereo image is problematic—but it is symmetrical.

Type 3—Stereo, optimized for the driver’s listening position. Levels and arrival times are tweaked to overcome Problem #1 and Problem #3, but only for one seat (the performance in the other seat is worsened). Some OEM systems also use asymmetric left/right EQ to address Problem #2. This is the most common high-performance system type sold by the aftermarket, and in those situations asymmetric EQ, level, and delay are all used regularly. There are very few vehicles sold with Driver-Optimized Stereo from the factory. Most OEM implementations that include driver-seat optimization make it a user-selectable setting in the tone-control menu (for example, Ford and Subaru have both recently offered this setting in specific vehicles).

Type 4—Stereo with phase manipulation to partially address Problem #3 to some degree, but for both front seats. A few years ago, phase manipulation was solely used in premium branded systems, but over the past few years phase manipulation has appeared in base audio systems from Toyota, Hyundai, Kia, and General Motors (possibly due to an intellectual-property lockout expiration?). It should be noted that Problem #1 and Problem #2 (near-side amplitude and near-side off-axis response) cannot be solved for both seats unless a speaker system is used, which can minimize the near-side asymmetry. For this reason, many premium OEM systems have been moving to the next category of presentation.

Type 5—Upmixed systems with a center speaker, delivering a symmetrical presentation from either side of the vehicle. These are very common now with premium vehicles, with Dolby Pro Logic II, DTS Neural, Harman Logic 7 and QLS, Bose Centerpoint and

Panaray, Fraunhofer Symphoria, and more all finding their way into vehicles from Cadillac and BMW, to Honda and Kia. While multichannel systems are often being labeled “immersive,” many implementations seem to include an upmixing algorithm for use with stereo sources (such as Bluetooth streamed from smartphones), and any height speakers are used in some application-specific manner (rather than playing object-oriented content).

The Challenges

As **Table 1** shows, there are pros and cons for each presentation type. Type 1 Stereo solves few of the acoustical problems in a vehicle cabin and is an endangered species in the OEM market. The presentation is symmetrical, even if it is not optimized. Type 2 Mono Center is more common than it used to be and is also symmetrical without being optimized. The center is largely in the center, but so are sounds that should not be, and comb filtering is significant. Some OEMs are shipping pseudo-upmixed systems, with frequency-band-limiting filters for the center speaker (rather than using an actual upmixing algorithm). Some premium OEM systems in the past few years have been found playing 200Hz and below in the doors, 2000Hz and above in the left and right dash locations, and the 200Hz to 2000Hz band is only played by the center midrange channel (which of course makes it mono—and terribly disappointing!). These often also delay the rears symmetrically.

Type 3 Delayed, Optimized for the Driver is rarely found in the OEM market, presumably due to a preference to symmetry on the part of vehicle manufacturers (who are the customers, even if they are not the end audience for the sound system). It should be noted that the pattern of comb-filter cancellations in either seat, caused by the left and right speaker path-length differences, can be completely eliminated in one seat with the proper use of delay, and this explains its appeal in the aftermarket. The aftermarket is currently focused on asymmetric performance in most cases. In general, aftermarket clients are seeking a higher level of performance than the OEM automaker is capable of delivering within the constraints of weight, current, and cost. While it is increasingly difficult to change an OEM in-dash receiver out for aftermarket, many aftermarket receivers now have level, delay, and even asymmetric EQ for driver-seat optimization (although one suspects these tools are rarely used effectively). These often also delay the rears symmetrically.

Type 4 Stereo with Phase Manipulation was initially only found in premium systems, but we have seen Toyota, Hyundai/Kia, and GM all offering

In-Car Stereo Presentations			
Presentation Type	Primary Technique	Symmetrical?	Trend
Stereo	None	Symmetrical	Declining
Stereo with mono center speaker	Summed center	Symmetrical	Declining
Stereo with driver’s optimization	Delay	Asymmetrical	Rare
Stereo with phase manipulation	All-pass filters	Symmetrical	Ascending
Upmixed systems	Upmixing algorithm	Symmetrical	Ascending

Table: 1 Aspects of the five stereo presentations.

base audio systems with all-pass filters in the output signal. It should be noted that in these base-audio applications, the comb filter pattern of cancellations in either front seat is only partially managed, but not completely eliminated. These often also delay the rears symmetrically.

Type 5 Multichannel Upmixed systems—whether or not they are also capable of playing multichannel recordings using technology such as Atmos—can deliver high-performance stereo performance to multiple seats. The center can be in the center (although it should be noted that it rarely is due to tuning decisions, apparently driven by preferences of the OEM automakers as customers). The algorithms used are usually the strongest part of the system—again, constraints on weight, current, and cost limit the audio performance of upmixed systems just as relentlessly as they do any other type of OEM audio system. One commonly found shortcoming is center speakers with poor performance and no high-frequency driver element (some center channels even eliminate the treble completely from the signal sent to the center-channel driver!) The aftermarket is learning to improve the performance of OEM systems while taking advantage of the OEM’s

licensed upmixing technology, and we hope to see high-performance aftermarket upmixed solutions available soon (at present only low-performance algorithms, unprotected by IP, are available in the vast majority of aftermarket products).

Conclusion

Before any general tuning technique can be adopted for the automotive environment—such as “apply some delay to the near side to move the image”—the crucial question of “one seat or two” must be answered. Then the tools and resources available must be determined. Only then can such elementary techniques as delay or level be appropriately used to support the desired result. 

About the Author

Ken Ward has more than 35 years of experience in consumer electronics—at the retail level, in technical support, as a product manager, as a manufacturers’ sales rep, and as a store owner. He started investigating and teaching OEM audio interfacing for cars 20 years ago and is a leading expert on OEM audio systems and how to interface with them. He has worked in the security, remote start, infotainment, telematics, and in-car computing fields, but audio is his first love. He joined Elettromedia in 2020 as Technical Marketing Manager.



GETTING THE MOST OUT OF
**VACUUM
TUBES**

By Robert B. Tomer

Vacuum Tubes—The Important 4/5ths of Tube Circuits

Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they've failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions . . . plus many, many more.

Like Watching a Play From Backstage

Author "Bud" Tomer, an international authority on tubes, is well-known for his candid "Tech Tip" bulletins and his slide-film presentations to service, military and engineering groups. Within this book you'll find a virtual encyclopedia on vacuum tubes—yet the author has not delved into mathematical details that would be of interest to only a few tube engineers. Everything is focused toward the interest of those who use tubes—either for replacement purposes or in original equipment designs.

In his refreshing, down-to-earth style, Mr. Tomer sheds light on many of the "Why so many tube types?" "What About Tube Testers?", etc.

Other Interesting and Informative Topics

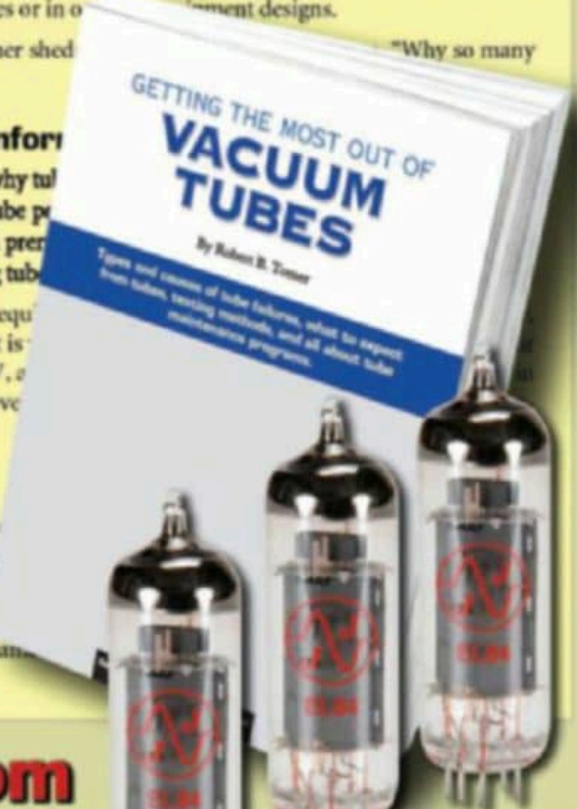
- 10 reasons why tubes fail
- Predicting tube performance
- Selected and preferred tube types
- Lengthening tube life

Anyone who has anything at all to do with tube equipment design, or proper use—should read this book. It is a must for anyone who designs, tests, maintains, and all about tube production and manufacturing processes—or even for hobbyists and radar gear.

You'll also learn about

- Catastrophic failures—what they are, when and how they happen
- Degenerative failures—what happens to tubes over time
- Subjective failures—why a seemingly normal tube may not perform
- Characteristic variables—differences that matter
- Special-purpose tubes—considerations for filament and thyratron types.

Order your copy
cc-webshop.com



A²B Audio Solutions

32+ channels of digital audio over twisted pair:

- up to 16 nodes
- 300m total length
- 50W of bus power

Reduce project time and risk

- Development & OEM modules
- Interface, ADC, DAC, Mic array boards, cables
- Fast turn custom designs



CLOCKWORKS
Signal Processing

www.clk.works info@clk.works
+1 978 258 5402

A²B is a trademark of Analog Devices Inc.

Using Multiphysics Simulation for an Electrovibroacoustic Analysis of a Balanced Armature Transducer

This article explains how multiphysics modeling and simulation can be used to analyze electroacoustic transducers. Using a balanced armature (BA) transducer as an example, the analysis combines the electromagnetic field with structural vibrations and acoustic radiation using multiphysics couplings.

By
Jinlan Huang, COMSOL

A balanced armature (BA) transducer, also called a balanced armature speaker (BAS), is a miniature loudspeaker commonly found in earbuds, hearing aids, and in-ear monitors (IEMs) due to its compact size and high performance [1]. It belongs to a group called moving-iron speakers, which is the earliest type of electric loudspeaker to have been invented. With a history that can be traced back to its application in the early telephones invented by Alexander Graham Bell, moving-iron drivers are electromagnetic (EM) transducers that rely on the magnetic forces (the Maxwell stresses) between magnetized bodies for motion.

Nowadays, BA transducers have evolved to become known for their small size, light weight, low distortion, and high electrical efficiency. A schematic representation of a modern transducer can be seen in **Figure 1**. It contains an armature (made of a magnetic alloy) that runs through a coil and is balanced between two permanent magnets. When an AC voltage is applied to the coil inducing an AC current, the arm is magnetized and alternating magnetic fluxes are generated. This makes up a magnetic circuit with a strong interaction between the AC magnetic field (generated by the coil) and the DC field of the permanent magnet. This in turn generates an alternating push-pull force on the armature. Since there is a driving rod/pin connecting the arm and diaphragm, the movement of the arm is passed onto the diaphragm, resulting in sound waves.

There is a growing trend to adopt multiple receivers in the same device, with each covering a specific target frequency band and working in combination to deliver the best overall sound. The pursuit of higher sound quality and smaller geometry size in each design requires a good understanding of the working mechanisms as well as many tests and trials, which is no different than the requirements of Bell's time.

However, with the development of modern technologies, engineers can now use numerical simulation to gain a better understanding of the physical phenomena involved, try out different ideas, and prototype with ease in a virtual environment. In the virtual world, one can freely change the structure, material, and operating conditions of a device or, if available, even use an optimization algorithm included in the simulation software to find an optimal design. In general, numerical simulations can be introduced at every stage of the design cycle, which often results in fewer physical prototypes; lower costs; faster time to market; and most importantly, better products. It can be extremely useful in the design of devices that involve multiple physical phenomena affecting one another, such as in balanced armature transducers.

When using simulations to evaluate a BA transducer's performance in a specific application, a lumped circuit representation of the transducer is often adopted to study its interaction with other

components in the system [2,3,4]. This lumped approach provides a quick and convenient way to perform a system analysis. However, when it comes to the design of the transducer or optimizing certain components for a specific frequency band, a full and detailed electrovibroacoustic analysis including all the acting electromagnetic and mechanical components of the transducer becomes necessary.

This type of detailed analysis provides an understanding of how some key design parameters affect the electromagnetic properties, the vibroacoustics behavior of the moving parts, the damping mechanisms, and the acoustic response. With a specific goal in mind, such a detailed model can also be used for an optimization analysis of certain driver components to determine the ideal dimension, shape, or topology and study the nonlinear distortions resulting from large deformations.

Now let's investigate how to set up a finite element model for such a detailed analysis [5]. The modeling example presented in this article is conducted in the frequency domain, assuming small perturbations and linearity under harmonic excitations.

Underlying Physics and Their Interplay

A full electrovibroacoustic analysis requires solving a fully coupled model that combines the electromagnetic field with structural vibrations and acoustic radiation. For the electromagnetic part of the simulation, Maxwell's equations are used to describe the magnetic flux in the air and the solids. However, due to the dynamics of the analysis, the magnetic and electric fields are coupled. Some of the most important electromagnetic quantities encountered in the analysis are the magnetic field intensity and flux density, the current density, the electric field and displacement, the electric conductivity, and the driving frequency.

The transducer contains different material types, and their electromagnetic behaviors can be described with various electric and magnetic constitutive relations, as well as electric conduction models. For example, you can prescribe a remanent flux density for the permanent magnetic, use a nonlinear B-H curve to describe the soft iron properties for the pole piece, and use a linear relation for other materials. In the coil domain, a homogenized current density can be prescribed.

Structural Vibrations

The vibrations in the structural domains are described by the dynamic force balance determined by Newton's second law. Some of the most important

mechanical quantities are the mass density; displacement field; stiffness; strain; and body load, such as the magnetic force induced inside the arm. The behavior of different solid materials can be described with different stress-strain constitutive relations. In this case, it is sufficient to use an isotropic linear elastic material model that assumes a linear relation between the stress and strain (i.e., the stiffness tensor is constant).

A nonlinear strain-displacement relation is used, which is necessary when the linear vibration model needs to be solved in a deformed geometry. This is required in this example because the armature experiences static Maxwell stresses, and therefore static deformation, in the permanent magnetic field. The static solution should be calculated first and then used as the linearization point for a frequency-domain perturbation study. This nonlinear formulation also allows for a large static deformation when the relation needs to be described in a deformed geometry.

Acoustics

For the acoustics in the air domains, special attention is needed in the slit-like narrow regions to capture the correct thermoviscous boundary layer losses. The acoustics in these areas are best described by the full set of linearized Navier-Stokes equations in quiescent background conditions (in other words, the governing equations describing the thermoviscous acoustics) [6]. For a frequency-domain analysis, the Helmholtz equation and an equivalent fluid model that adds or smears the boundary layer losses onto the bulk of the fluid are used. This is because analytical expressions exist

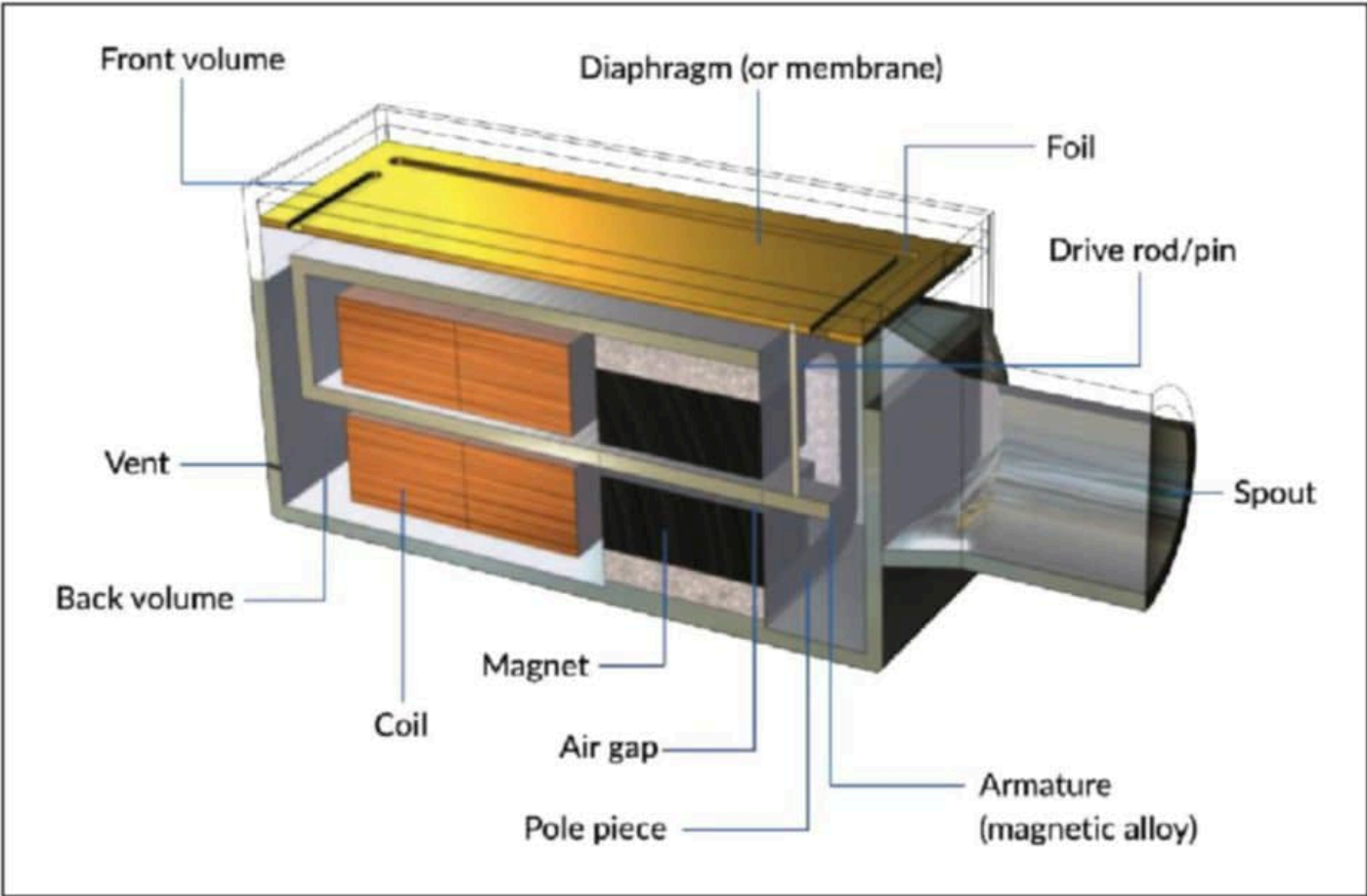


Figure 1: The components of the example balanced armature transducer (from Reference 1).

for many geometries where the losses associated with the acoustic boundary layers are introduced through frequency-dependent, complex-valued speed of sound and density. These are known as low-reduced-frequency (LRF) models [7,8,9] and are less computationally expensive compared to a full, detailed thermoviscous acoustic model. Some of the most important acoustical quantities are the acoustic pressure, fluid density, speed of sound, and driving frequency. In the narrow regions

Table 1: Characteristics of the example balanced armature transducer.

Dimensions of the transducer box	4mm×2.15mm×2.8mm
Spout's inner diameter	1mm
Diaphragm's thickness	50μm
Armature's thickness	100μm

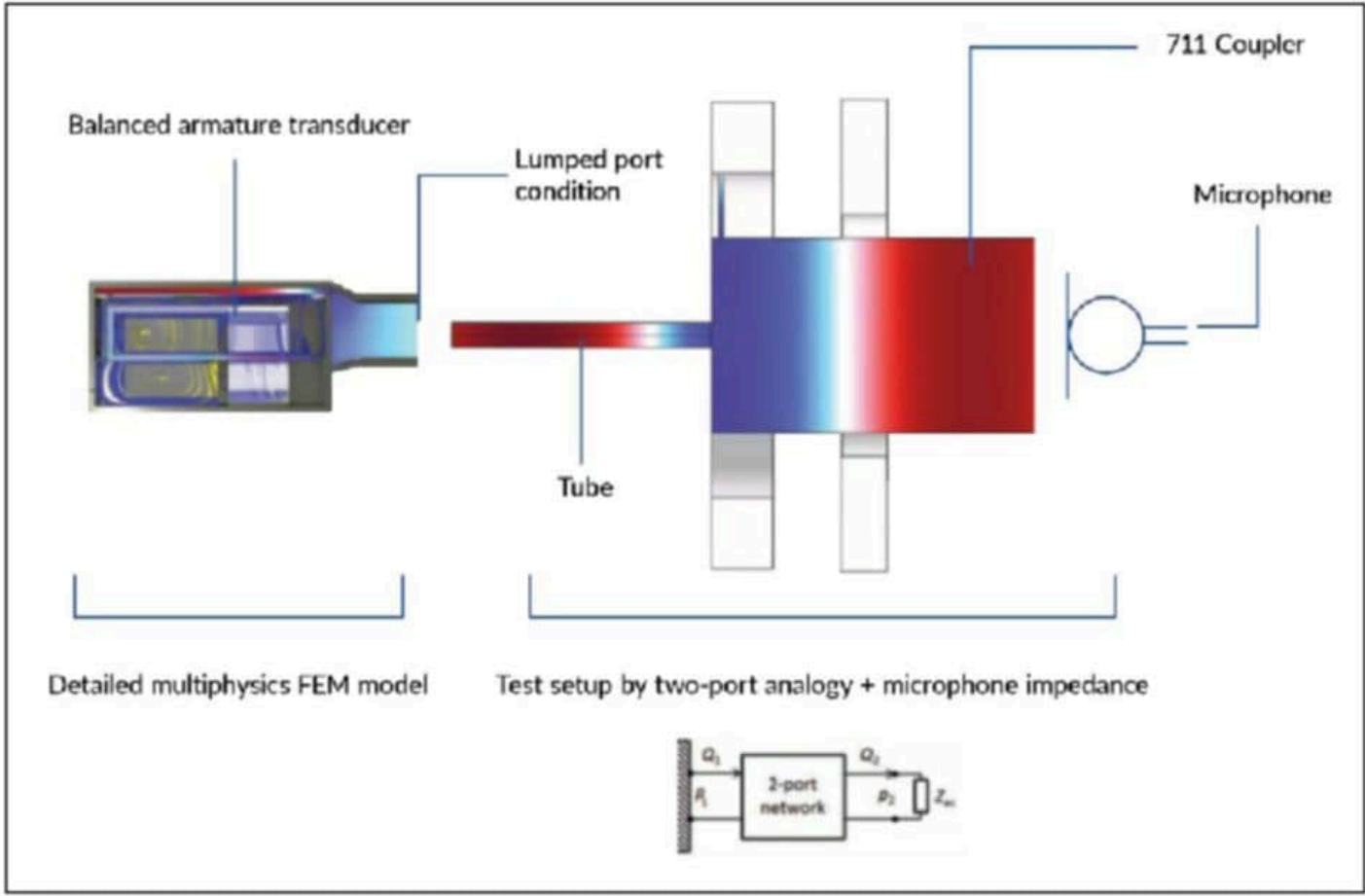


Figure 2: A common measurement setup for testing the balanced armature speaker used in a hearing aid (from Reference 1).

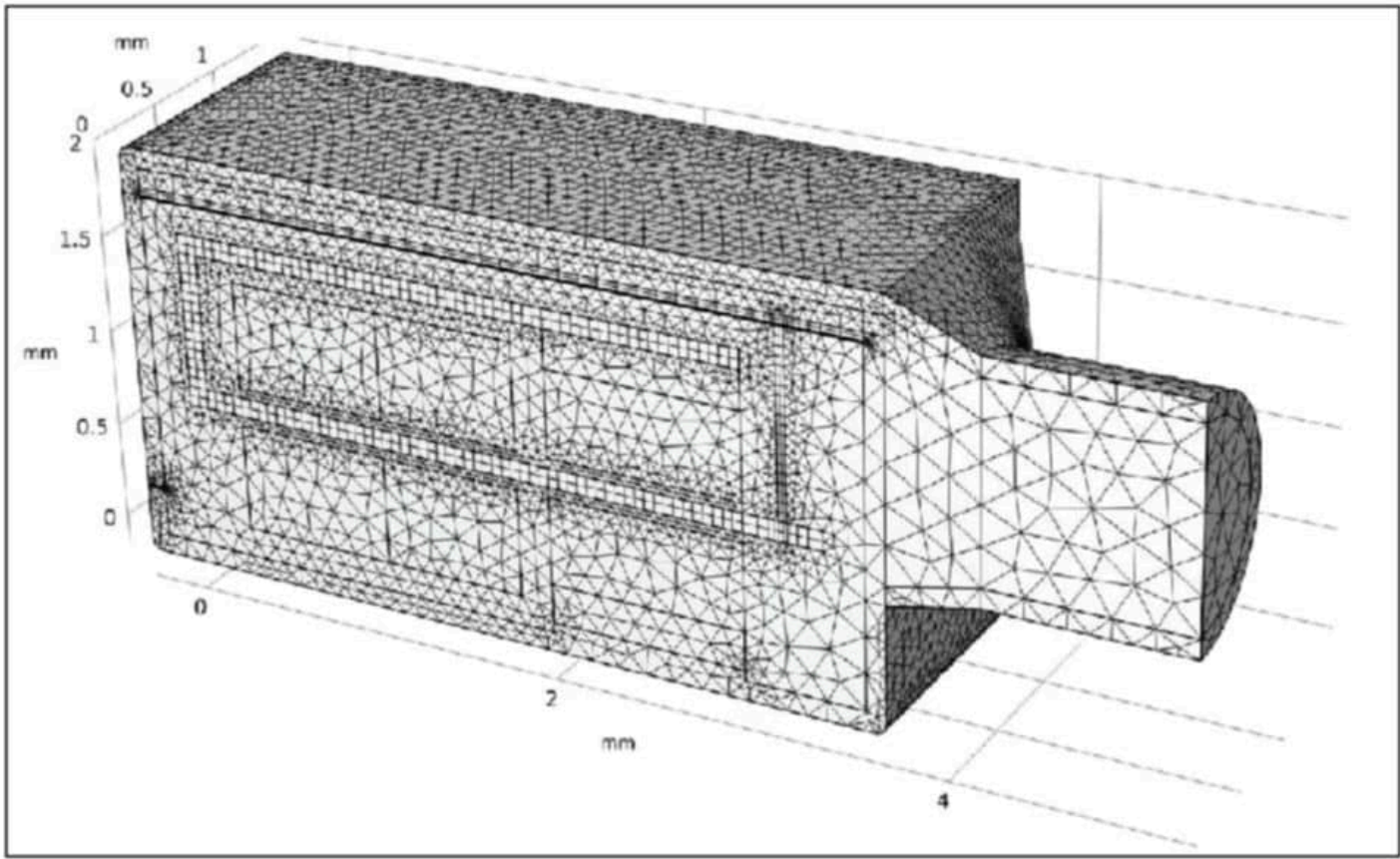


Figure 3: The mesh used in the frequency-domain analysis.

where an LRF model is used, the density and speed of sound become complex-valued quantities to model the desired losses.

Interaction Between Deformable Solids and Magnetic Fields

When magnetized by an AC current applied to the coil, the armature lying in the air gap experiences the Maxwell stress that forces it to move. This stress includes two contributions: one is volumetric, accounting for the stress that is present inside the magnetized bodies, and the other is on the surfaces, accounting for the stress that comes from the interactions with the surrounding magnetic field.

In addition, the solid deformation influences the material magnetization, which can be included by taking the EM vector components on the material frame [10]. Furthermore, the varying air gap introduces a change in the magnetic flux circuit (i.e., a varying reluctance), and this effect can be captured by modeling the geometric change in the air domain. Therefore, the magnetic model should be solved in a deformed geometry to correctly simulate the effects from solid movement (the back electromotive force, or the back EMF).

Interaction Between Solid Vibrations and Acoustics

The vibrating structures cause the surrounding fluids to move back and forth and thus create sound. They also experience the pressure load from the oscillating fluid, which in turn affects their vibrational behavior.

When the Helmholtz equation is solved for acoustics, the acoustic–structure coupling can be achieved by using the structure’s normal acceleration as an excitation for acoustic radiation and applying acoustic pressure as a boundary load on the vibrating structure. In this case, the continuity at the fluid–solid interfaces is enforced only in the surface normal (similar to a slip condition) because a Helmholtz model does not resolve the boundary layers in detail.

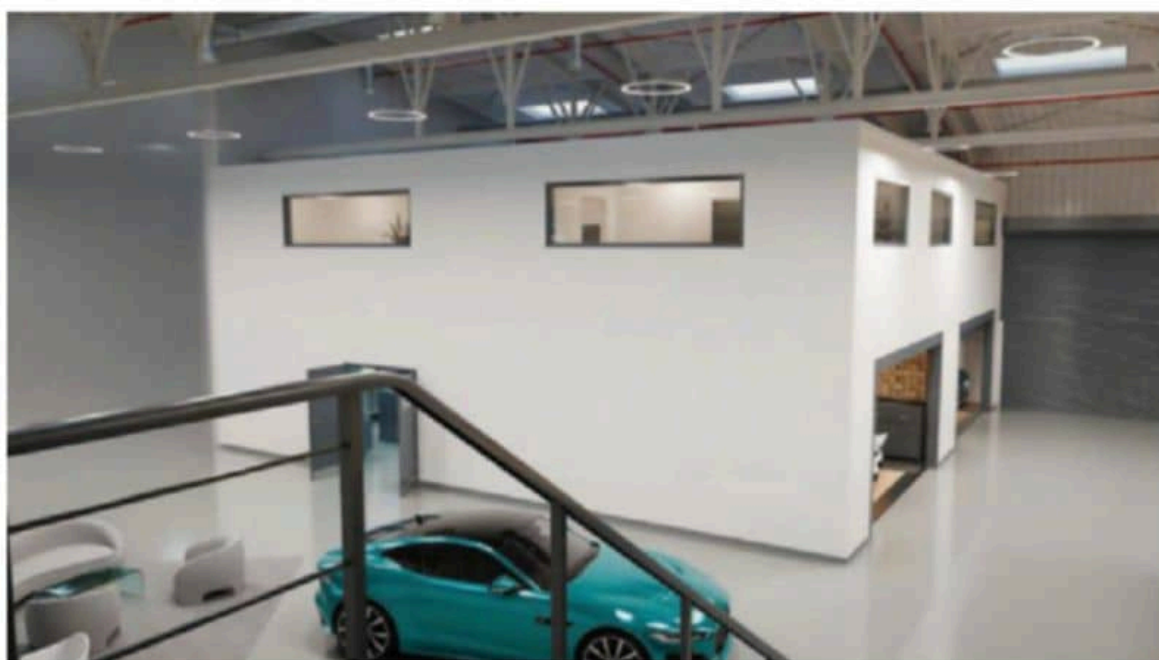
Setting Up the Multiphysics Model

We solved the fully coupled model in the COMSOL Multiphysics software using general-purpose geometry and material properties for the balanced armature transducer [1,5] (COMSOL Multiphysics is a registered trademark of COMSOL AB.) **Table 1** shows some characteristic dimensions of the transducer.

The material data for this example is from the Material Library in COMSOL Multiphysics. Steel is used for the driving rod and diaphragm; a soft iron with losses is used for the pole piece; and a generic

Creating a new paradigm for automotive audio

Audio Foundry is the first-of-its-kind social workspace and collaborative makerspace designed exclusively for advanced automotive audio ideation, prototyping, benchmarking, and advanced research. This unique, members-only hub provides access to cutting-edge tools and premium Peerless audio components.

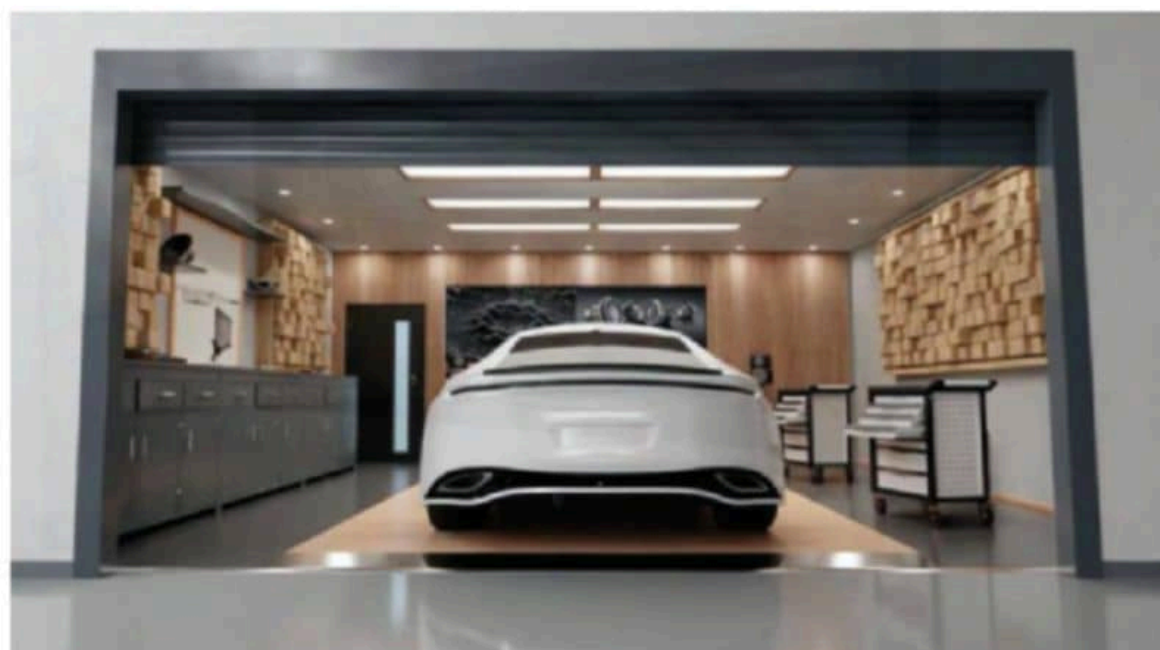


The Makerspace

Fully equipped, state-of-the-art location in Wales, UK

The Rig

An SUV-sized development sandbox for acoustic design and system layout



The future of automotive audio starts here

Uniting the best minds to pioneer a new era in car audio

Contact us at audio-foundry.com to speak with an Audio Foundry specialist.

steel with a relative permeability of 1000 is used for the armature. The coil has a voltage of 0.2V (RMS) and is made from 1000 turns of a copper wire with a diameter of 32 μ m. The permanent magnet has a remanent flux density of 0.4T. (See Reference 1 for detailed information about this model.)

The transducer model as well as the full test setup can be seen in **Figure 2**. The setup includes an earmold tube that is 10mm long and has a radius of 0.5mm. The 711 coupler, or occluded-ear simulator, is a generic design that meets the IEC 60318-4 international standard [11]. A measurement microphone (which has representative RCL values of $R = 119\text{e}6 \text{ Ns/m}^5$, $C = 0.62\text{e-}13 \text{ m}^5/\text{N}$, and $L = 710 \text{ kg/m}^4$) is used to terminate the coupler. To simplify the modeling approach, the test setup is defined with a transfer matrix and is coupled

with the transducer via the Lumped Port feature in COMSOL Multiphysics. More information on the 711 coupler and how to compute the transfer matrix of a tube and coupler measurement test setup can be found on the COMSOL website [12,13], respectively.

COMSOL Multiphysics provides built-in interfaces covering a range of application areas, including acoustics, structural mechanics, and electromagnetics. These interfaces define sets of equations, boundary conditions, initial conditions, predefined meshes, predefined studies with solver settings, and predefined plots and derived values. In addition, there are several multiphysics interfaces available for combining physics and modeling multiphysics interactions, such as electromagnetic forces stemming from a magnetic-mechanical coupling.

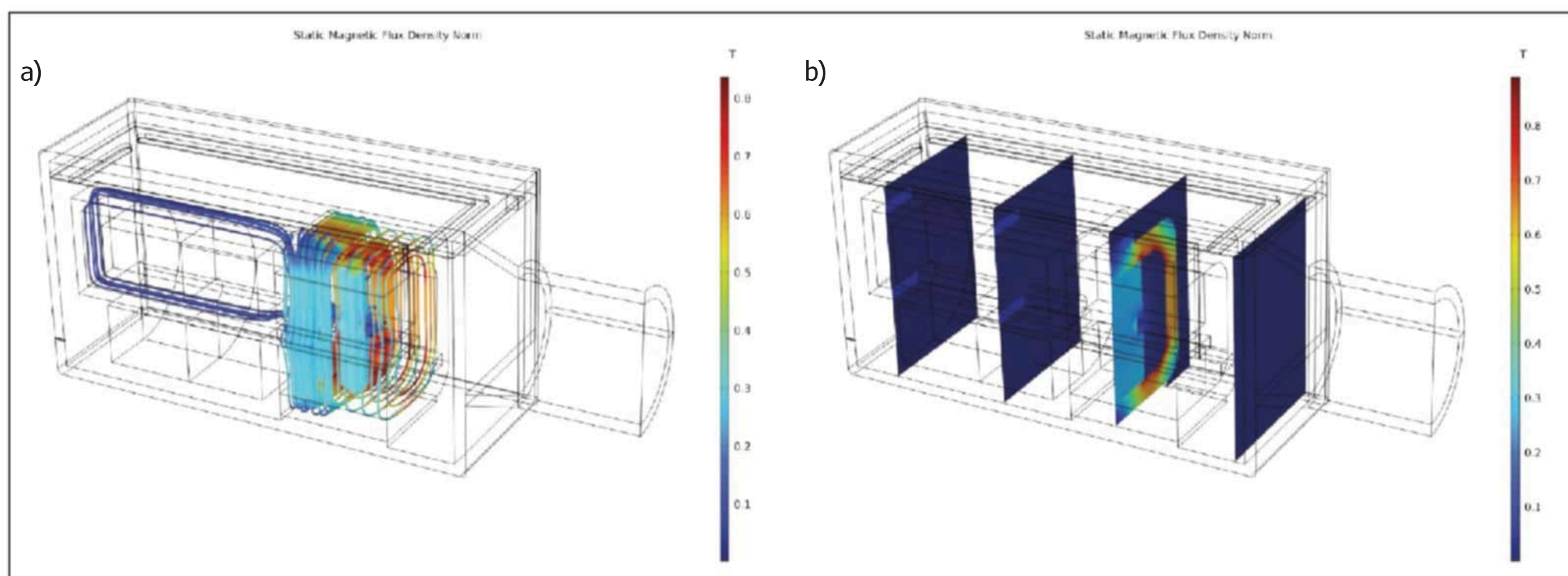


Figure 4: The DC magnetic field generated from the permanent magnet. The magnetic flux density (the B field) using streamlines (a) and its magnitude in slices (b).

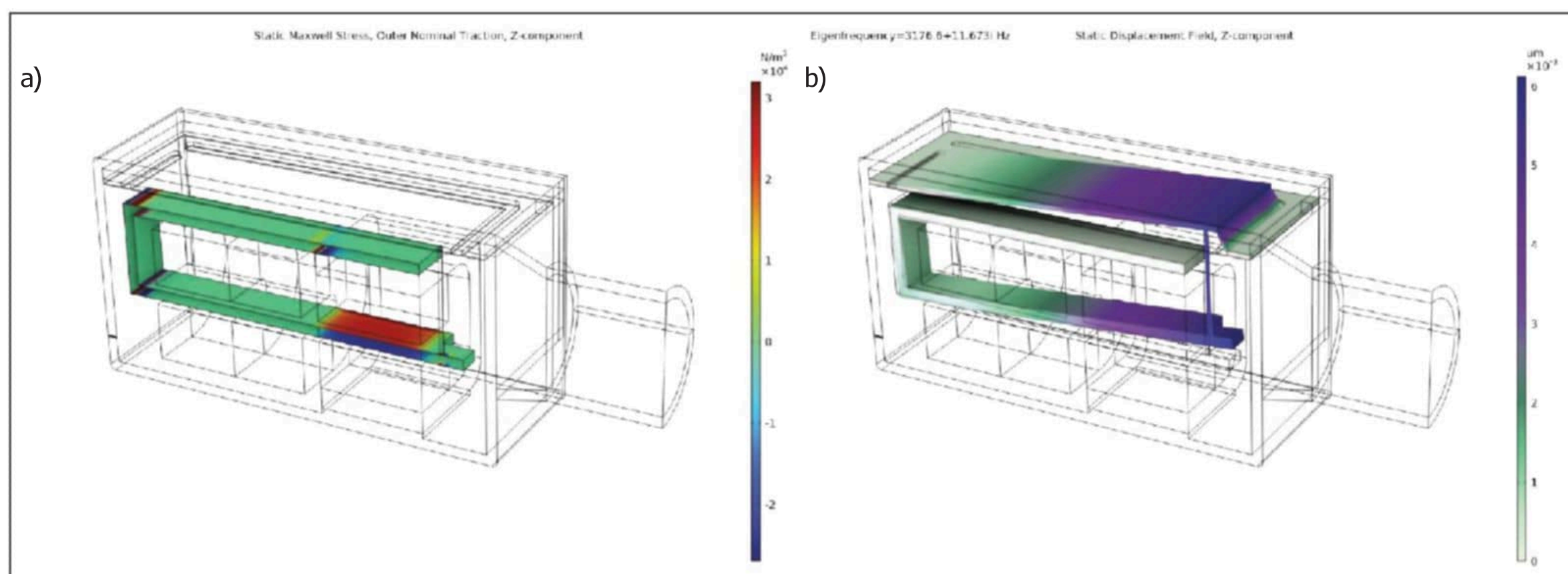


Figure 5: The Z-component of the Maxwell stress exerted on the armature (a) and the displacement of the moving components (b) induced by the net upward force.

For this example, the Magnetomechanics multiphysics interface is used, which combines Solid Mechanics, Magnetic Fields, and Moving Mesh interfaces. The Solid Mechanics and Magnetic Fields interfaces are coupled in the armature domain via the Magnetomechanical Forces coupling feature, which calculates the Maxwell stress exerted on the armature as well as the effect of its deformation on material magnetization. The Moving Mesh interface allows for defining the air domains next to vibrating structures as deforming domains where the mesh can deform with the solid movements. This is necessary to capture the back EMF.

The frequency-domain analysis correlates to a small-signal (or linear perturbation) analysis where linearized governing equations are solved. Since the model includes a permanent magnetic, and thus a DC field, the Maxwell's equations are linearized about the DC component, and the DC "balanced" location of the armature is used as the linearization point for the structure. In COMSOL Multiphysics, this type of analysis is known as Frequency Domain Perturbation. All the AC (perturbation) fields in this example, including the acoustic field, are assumed to be time harmonic.

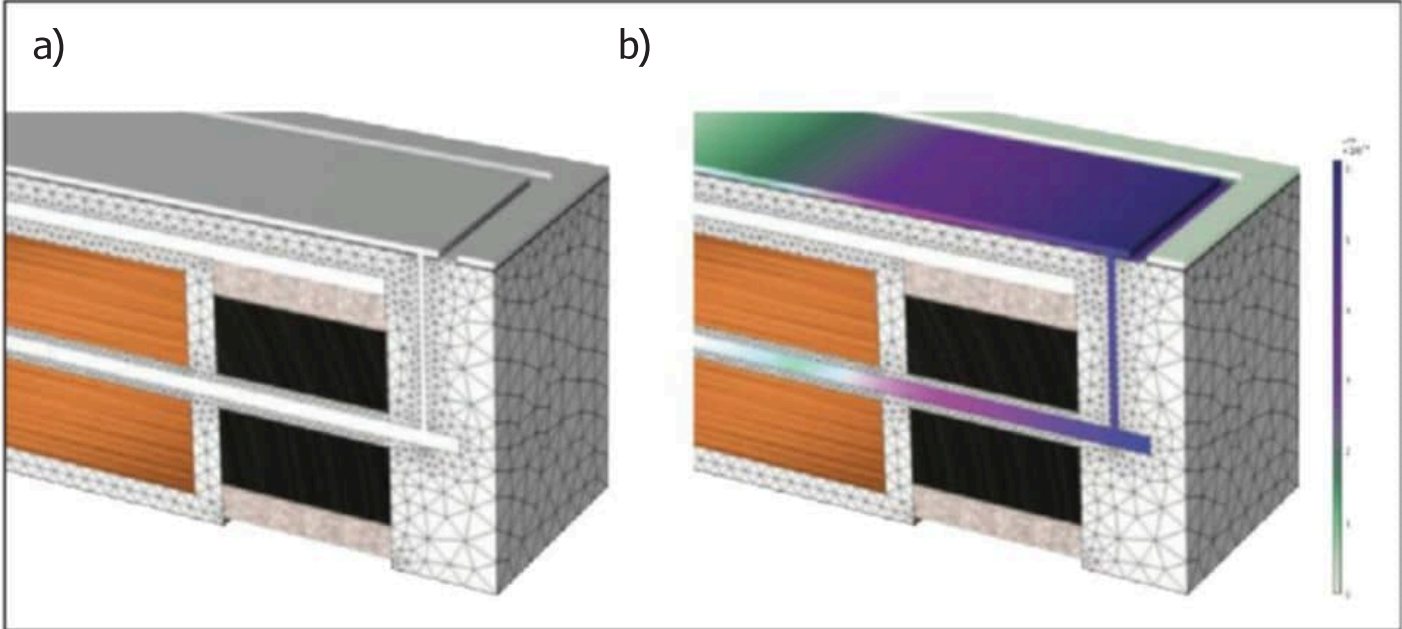


Figure 6: A close-up of the mesh around the largest displacements after the static actuation (b), as compared to the original mesh (a).

About the Author

Jinlan Huang currently works as a Principal Applications Engineer at COMSOL in Burlington, MA, with a specialty in vibrations and acoustics. She has been a speaker of multiphysics modeling in the fields of Acoustics, Structural Mechanics, MEMS, Fluid Flow, and Heat Transfer and an instructor of specialized courses in Acoustics. She received her PhD from Boston University, Department of Aerospace and Mechanical Engineering, investigating acoustic wave propagation in complex-tissue environments and ultrasound-induced tissue heating and bleeding control. She joined COMSOL in 2011.



The QA403 Audio Analyzer...\$599

-110 dB THD+N
+18 dBV Output (4 ranges)
+32 dBV Input (8 ranges)

USB Powered
Fast, automated testing
REST Programmable



www.QuantAsylum.com
Designed and built in the State of Colorado USA

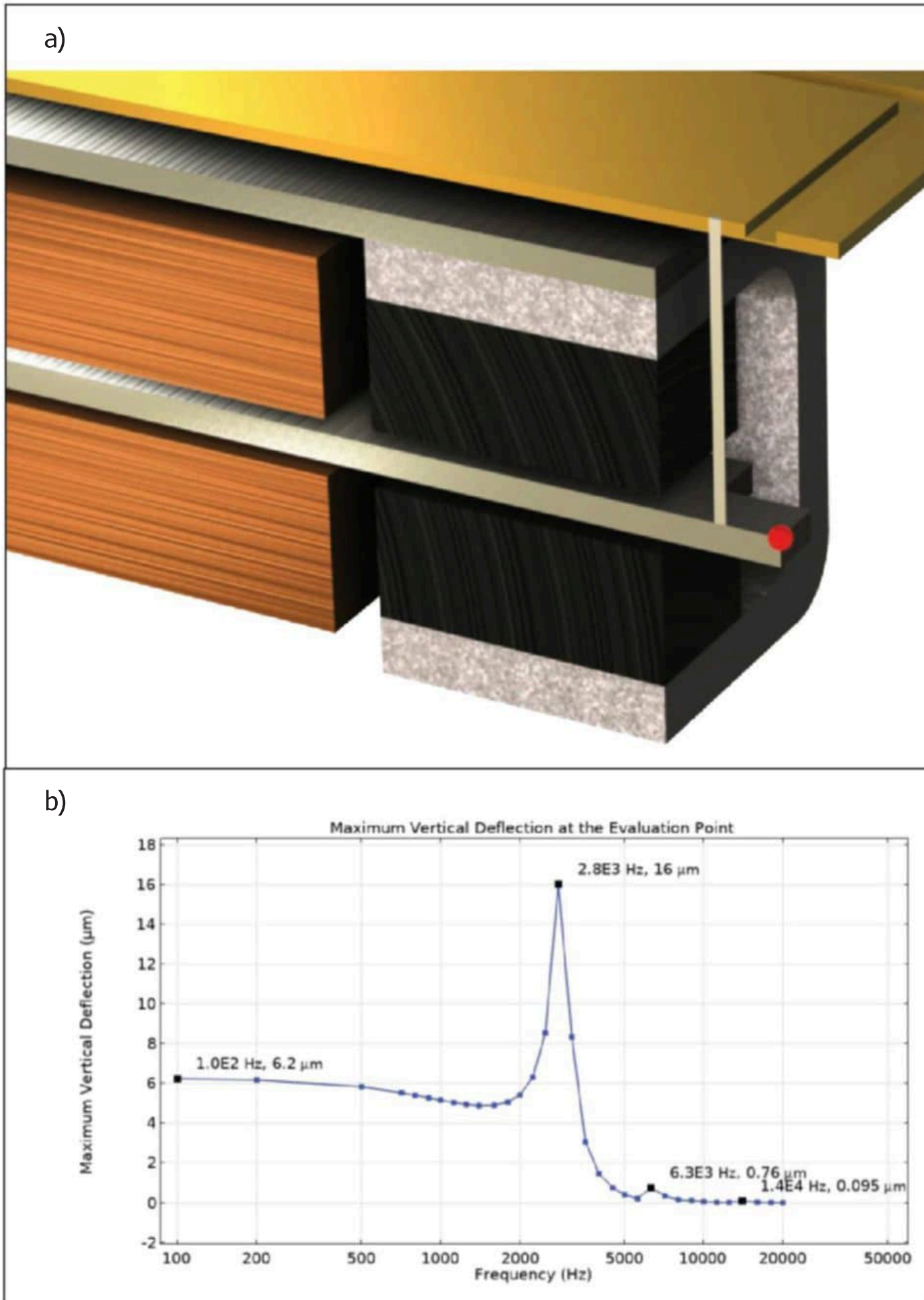


Figure 7: Maximum vertical displacement at the evaluation point, which is the red point shown in (a) and as a function of driving frequency (b).

For the acoustic field, we use the Pressure Acoustics, Frequency Domain interface that solves the Helmholtz equation. The front volume, the spout, and the vent areas are modeled using the Narrow Region Acoustics feature, an equivalent fluid model, to introduce the thermoviscous boundary layer losses in those regions. The acoustic domains are connected to the vibrating structures through the Acoustic-Structure multiphysics coupling, which accounts for the fluid load on the structure as well as how the structural acceleration excites the fluid.

Note that in this model the casing of the transducer is assumed rigid and fixed. It is also possible to make the casing elastic, and in this way compute the external forces and torque that the transducer exerts on its surroundings. The model is solved over a frequency range with the single mesh depicted in **Figure 3**.

Results and Discussion

DC Field Solutions — Let's first look at the DC solutions generated by the permanent magnet. **Figure 4** shows the magnetic flux density (the B field) using streamlines (a) and its magnitude in slices (b). This DC magnetic field imposes Maxwell stresses on the armature, resulting in a static displacement of the armature and other moving components that are connected to it.

As seen in **Figure 5**, the portion of the armature sitting inside the air gap experiences the strongest traction (a), with a net force pointing upward, which pulls the solids so that they move upward (b). Note that a scale factor of 150 is applied to the deformation plot for visualization purposes.

The structural deformation directly affects the air-domain geometry, and this change is captured by the moving mesh functionality. **Figure 6** shows a close-up of the original mesh when the armature is at rest (a) and the deformed mesh when it is actuated by the static magnetic force (b).

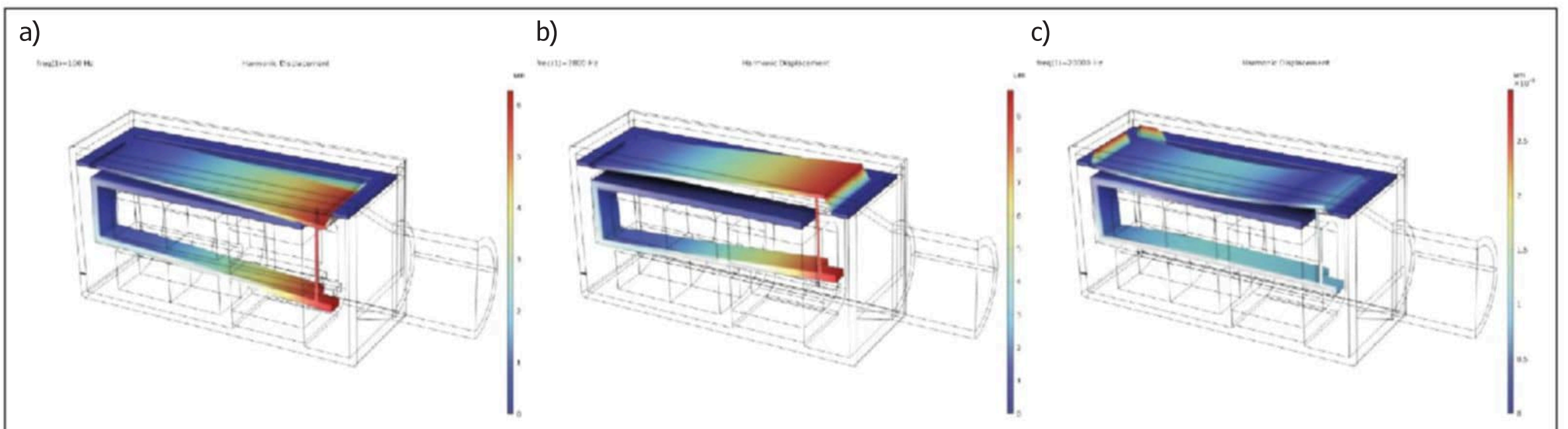


Figure 8: The vertical displacement field at 100Hz (a), 2800Hz (b), and 20kHz (c).

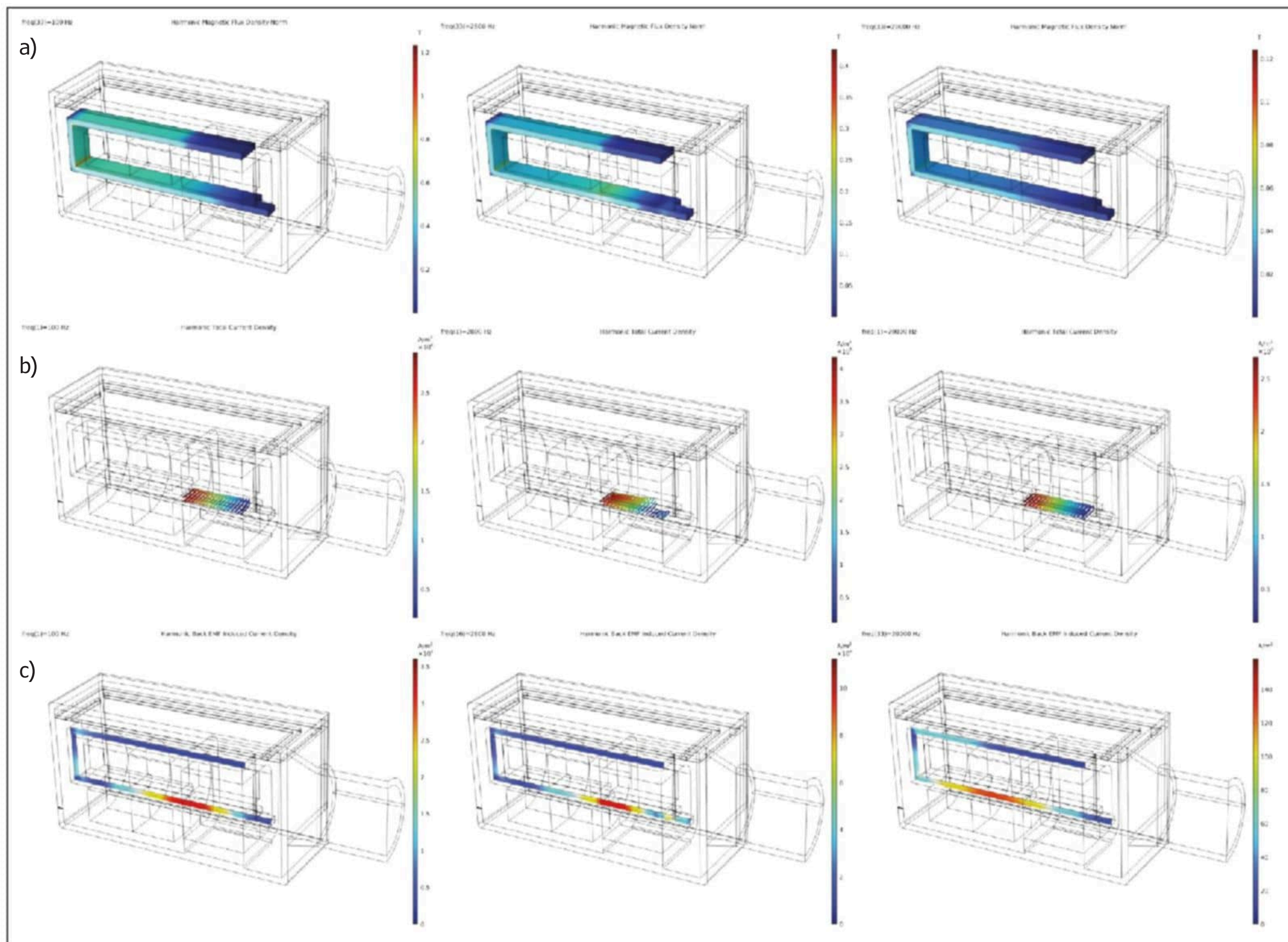


Figure 9: Harmonic EM fields in the armature for 100Hz, 2800Hz, and 20kHz showing the magnetic flux density (a), the total current density (b), and the back-EMF-induced current density (c).

References

- [1] Balanced Armature Transducer - Frequency Domain Analysis, COMSOL, www.comsol.com/model/balanced-armature-transducer-frequency-domain-analysis-61741
- [2] M.R. Bai, B.C. You, Y.Y. Lo, "Electroacoustic analysis, design, and implementation of a small balanced armature speaker," *Journal of the Acoustical Society of America*, Vol. 136, No. 5, pp. 2554–2560, 2014.
- [3] J. Jensen, "Nonlinear Distortion Mechanisms and Efficiency of Miniature Balanced Armature Loudspeaker," PhD thesis, DTU Electrical Engineering, 2014.
- [4] R. Christensen, "Lumped Element Modeling of Transducers," *audioXpress*, March 2023.
- [5] M. Herring Jensen, "Full Multiphysics Electro-Vibroacoustic Analysis of a Balanced Armature Transducer," INTER-NOISE and NOISE-CON Congress and Conference Proceedings (Inter-Noise 2022), Institute of Noise Control Engineering, pp. 1685–1691, 2022, www.ingentaconnect.com/contentone/ince/incecp/2023/00000265/00000006/art00078
- [6] M. Herring Jensen, "Theory of Thermoviscous Acoustics: Thermal and Viscous Losses," COMSOL, 2014, www.comsol.com/blogs/theory-of-thermoviscous-acoustics-thermal-and-viscous-losses
- [7] R. Kampinga, "Viscothermal Acoustics Using Finite Elements Analysis Tools for Engineers," PhD thesis, University of Twente, the Netherlands, 2010.
- [8] M.R. Stinson, "The propagation of plane sound waves in narrow and wide circular tubes, and generalization to uniform tubes of arbitrary cross-sectional shapes," *Journal of the Acoustical Society of America*, Vol. 89, p. 550, 1990.
- [9] H. Tijdeman, "On the propagation of sound waves in cylindrical tubes," *Journal of Sound and Vibration*, Vol. 39, p. 1, 1975.
- [10] J. Huang, "Modeling Speaker Drivers: Which Coupling to Use," COMSOL, 2022, www.comsol.com/blogs/modeling-speaker-drivers-which-coupling-feature-to-use
- [11] IEC 60318-4:2010, Electroacoustics – Simulators of human head and ear, Part 4: Occluded-ear simulator for the measurement of earphones coupled to the ear by means of ear inserts, IEC, edition 1.0, 2010.
- [12] Generic 711 Coupler — An Occluded Ear-Canal Simulator, COMSOL tutorial model, www.comsol.com/model/generic-711-coupler-8212-an-occluded-ear-canal-simulator-12227
- [13] Transfer Matrix of a Tube and Coupler Measurement Setup, COMSOL, www.comsol.com/model/transfer-matrix-of-a-tube-and-coupler-measurement-setup-104291

A scale factor of 25 is used here for showing the deformation. Note that the magnetic field is solved in this deformed geometry, which, together with the DC magnetic field and static solid displacement, serves as the linearization point for the frequency-domain perturbation analysis.

Frequency-Domain Solutions — Now let's look at the harmonic perturbation solutions on top of the DC bias. The study sweeps over a wide frequency range from 100Hz to 20kHz, and a plot of the maximum vertical deflection versus frequency at an evaluating point (located at the edge of the arm, as indicated by a red dot in **Figure 7a**) gives a good picture of the transducer's behavior in regard to the driving frequency (**Figure 7b**; the blue point marker in the curve indicates the frequencies being studied). Here, we see a maximum displacement of 16μm at 2800Hz and two smaller peaks at 6300Hz and 14,000Hz. The vertical displacement field in all moving parts is depicted for 100Hz (a), 2800Hz (b), and 20kHz (c) in **Figure 8**, which shows that higher-order modes are excited in the diaphragm for higher frequencies (the displacement is exaggerated).

Figure 9 displays the harmonic EM field inside the armature induced by the AC coil current at

100Hz, 2800Hz, and 20kHz, respectively. The results show that the magnitude of the B field (a) generated inside the arm decreases with frequency, while the magnitude of the total current density (b) as well as the back-EMF-induced current density (c) reaches the highest value in the air gap at 2800Hz. This is because the back EMF is strongest at the resonance when the structure deforms the most and thus the air gap in the magnetic flux varies the most.

The effect of the resonance is also clearly seen in the electric impedance plot of the coil, as illustrated in **Figure 10**. The two solid lines show the absolute value of the coil impedance (blue) and its phase (red) calculated from our full multiphysics model.

In lower frequencies, the impedance comes mainly from the resistive part, and its value as the frequency goes to 0Hz gives the DC resistance, which has an approximate value of 90Ω. At 2800Hz, when mechanical resonance occurs, the reactive part of the impedance changes its behavior from inductive to capacitive.

After the resonance frequency, the inductance of the voice coil starts playing a more important role, resulting in a continued increase in the coil impedance. Here, we also include the published data (the two dashed lines) obtained from the lumped model by René Christensen of Acculution A/S, a modeling and simulation consultancy [4] for comparison. There is good agreement up to a frequency where the lumped model is not applicable anymore.

While the location of the resonance matches well, we see the biggest difference in the magnitude around the resonance. This is expected because the full FEM model includes (and a lumped model does not) the effect of the geometry change (in both the solid and fluid domains), which is the strongest around the resonance. Moreover, the magnitude at the resonance is solely controlled by damping. A slight difference in the damping description can result in quite notable discrepancy in the magnitude, and that is why one should be careful in comparing the response at and near the resonance.

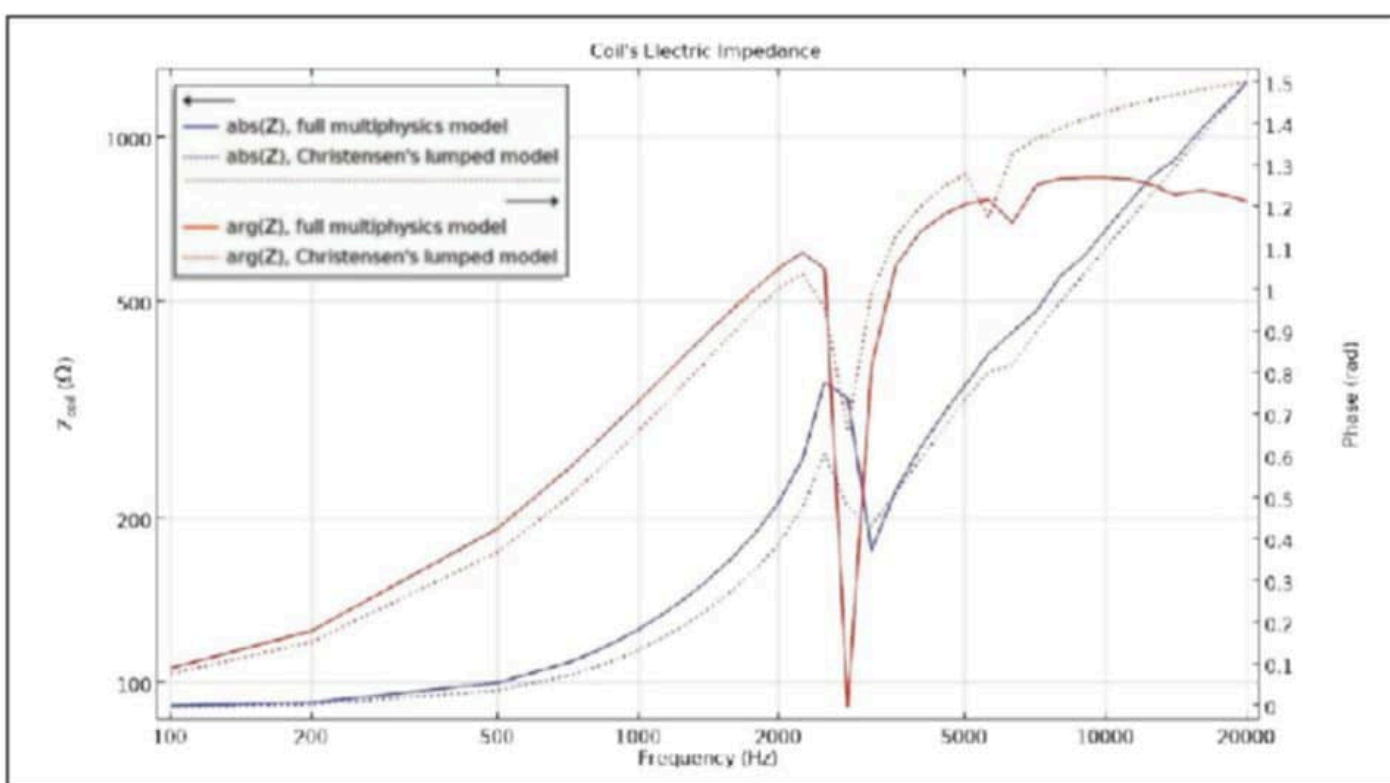


Figure 10: The electric impedance of the coil as a function of frequency. (Lumped parameter curve data courtesy of René Christensen, Acculution A/S)

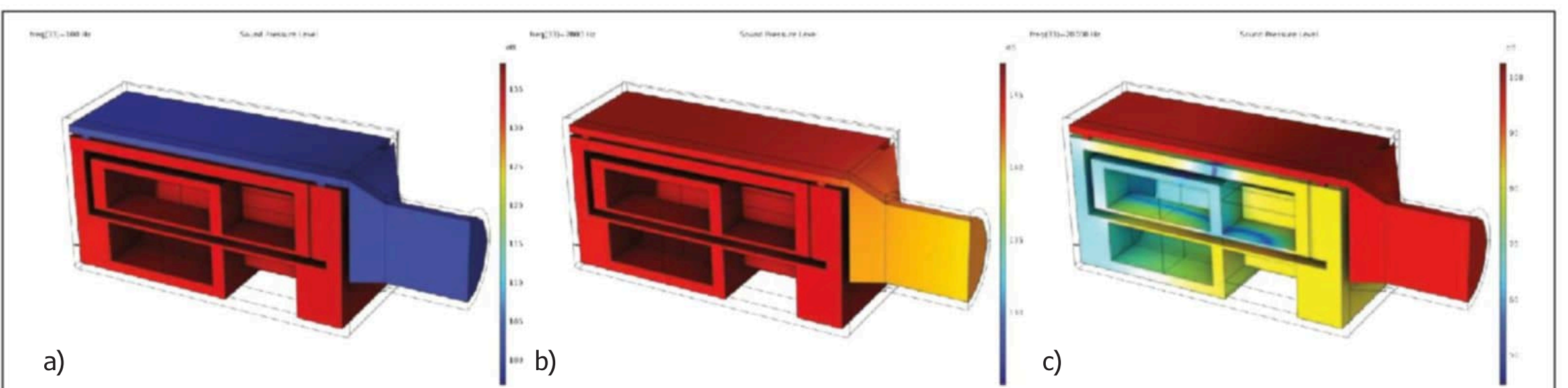


Figure 11: Sound pressure level at (from left to right) 100Hz (a), 2800Hz (b), and 20kHz (c).

Finally, let's look at the acoustic response. The three images in **Figure 11** show the sound pressure level at 100Hz (a), 2800Hz (b), and 20kHz (c), respectively. We can see that the sound pressure level is pretty evenly distributed on the cross section of the spout for all frequencies, and a maximum value of 142dB is obtained on the outlet at 2800Hz.

Figure 12 presents the sound pressure level received by the microphone that is used to measure the coupler response (Figure 2). The three resonances we see here coincide with those indicated in the displacement response curve shown in Figure 7. The plot also includes the results predicted from Christensen's lumped model (red curve). Again, the agreement is quite good.

Conclusion

Now, we have discussed how to conduct a full electrovibroacoustic simulation of a balanced armature transducer. The analysis combines the electromagnetic field with structural vibrations and acoustic radiation, including the interplay between the multiple physical phenomena. A moving mesh is used to model the back EMF so that the varying reluctance of the air gap in the magnetic circuit is captured. The frequency-domain

study gives detailed information for understanding and evaluating the performance of the driver undergoing small perturbations, which is important in the development of this type of transducer. The detailed model presented here can be used in an optimization analysis for innovative designs. 

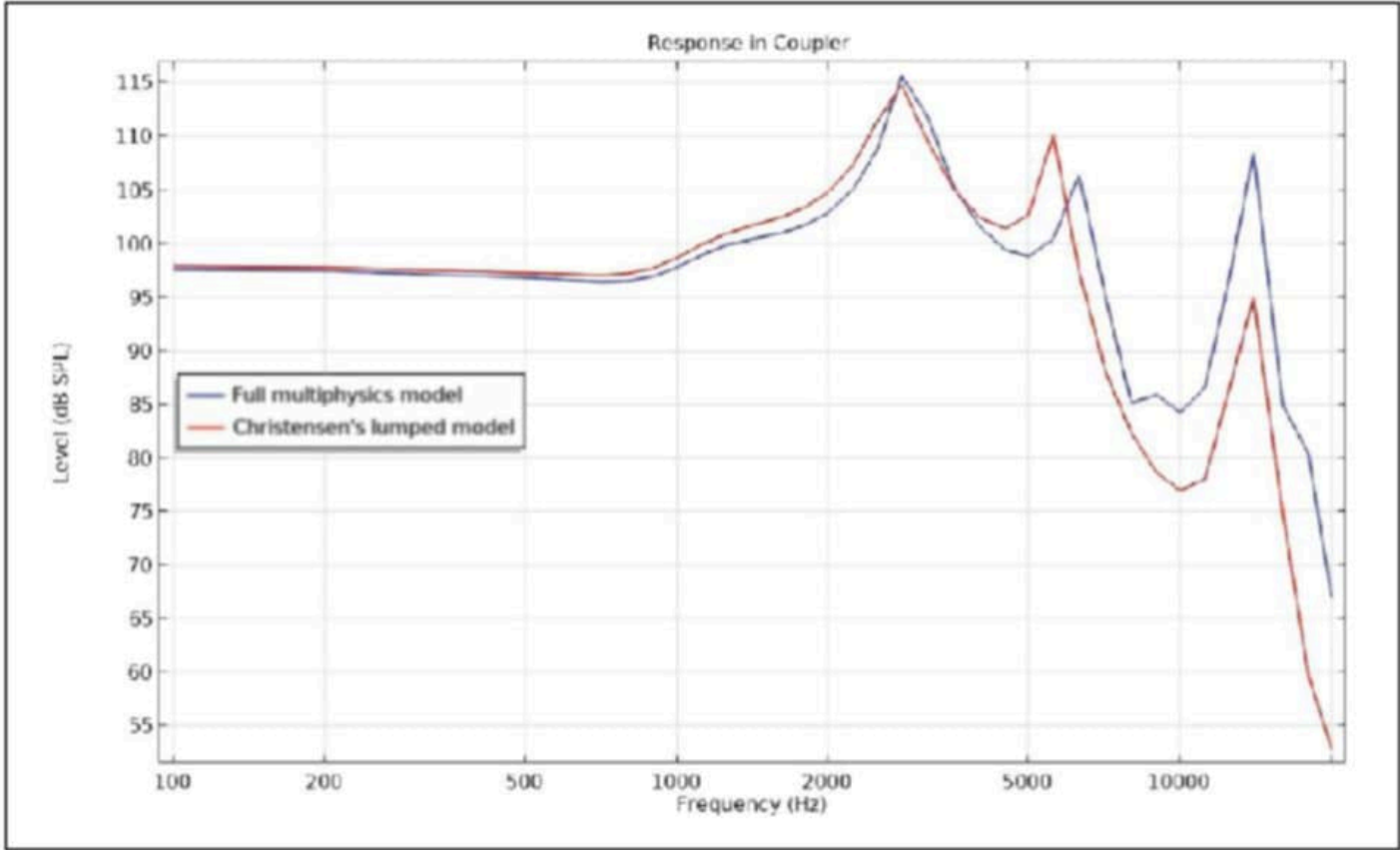
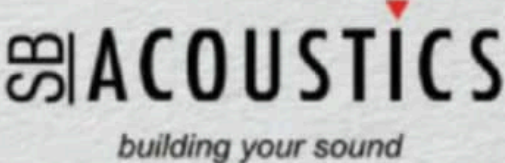


Figure 12: Sound pressure level detected by the measuring microphone as a function of frequency. (Lumped parameter curve data courtesy of René Christensen, Acculution A/S)



Gema

is a large 3-way speaker paying homage to the classic speakers of the 1970's. Generous enclosure proportions fitted with large bass drivers were the bee's knees at the time. And for good reason!

The Gema is our take on a retro-styled speaker, and the much loved 12 inch SB34NRXL75-8 lays the foundation of a deep, well controlled and tuneful bassline.



Scan the code to find your nearest distributor



Find us on:

 SB Acoustics

 sbacoustics

 SB Acoustics

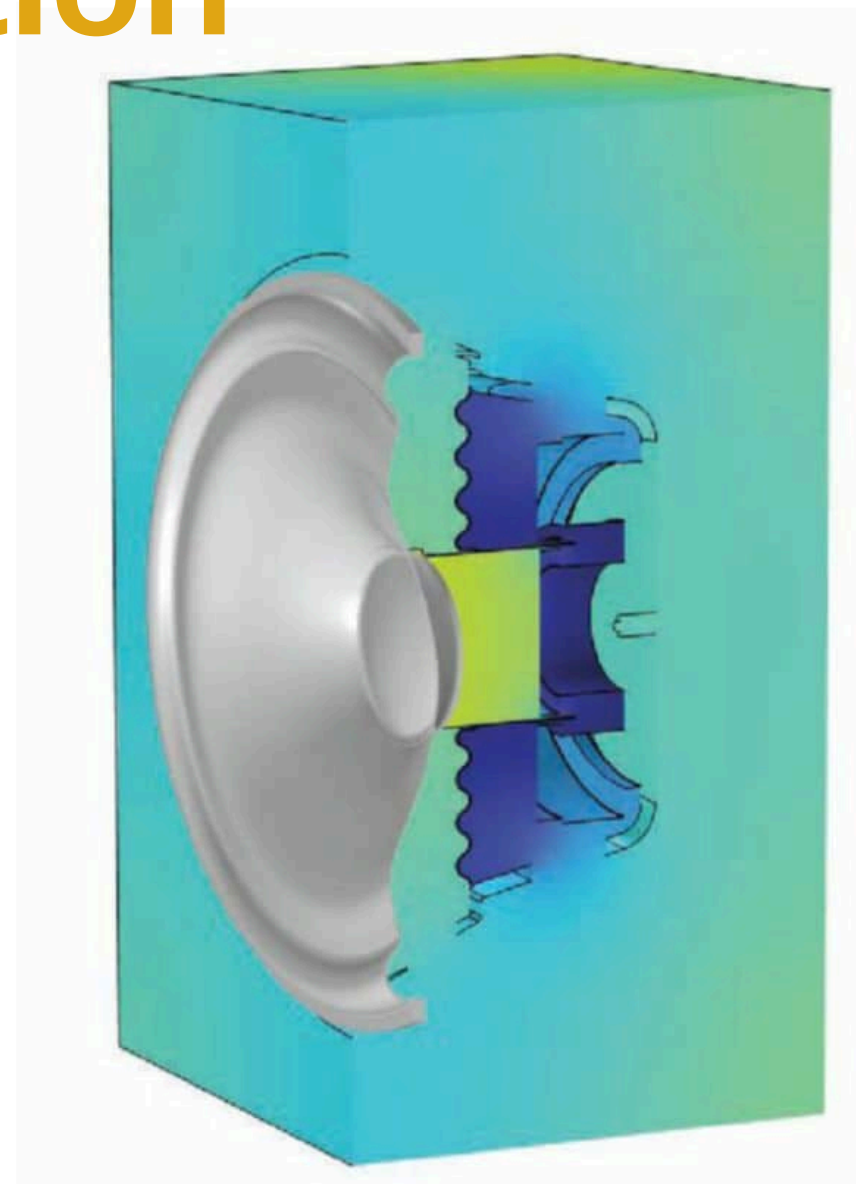
 sbacoustics.com

audioxpress.com | June 2024 | 45

Dimensional Reduction for Vibroacoustical Simulations

Computational demands can be very high when doing simulations involving coupled physics, such as for electromagnetics, structural mechanics, and acoustics—as is often the case for vibroacoustical transducers. This article looks at how to reduce the computational demands and associated costs utilizing symmetry schemes and dimensional reduction techniques, and the potential associated issues.

By
René Christensen



Any remedies to lessen this computational load can improve the overall workflow, and design changes can be evaluated faster, so it is always a good idea to seek ways to simplify model complexities. Here, we will have a look at how to reduce computational costs utilizing symmetry schemes and dimensional reductions techniques, and potential issues associated with them.

All aspects of any transducer or any other product are essentially three-dimensional, and so to get the full picture of its operation, any simulation should essentially be carried out utilizing the full geometry. However, in some instances symmetry conditions can be invoked to reduce the total number of degrees-of-freedom in the computational setup. There

are several considerations regarding whether the symmetry in geometry, boundary conditions, load/sources, and/or field, or only in a subset of these.

An example of this is shown in **Figure 1**, where a geometry has a symmetry plane, but applying a symmetry or “mirror” condition on the midline and solving this as an eigenmode problem using only half the geometry in the simulation would not return all possible modes, as some will have anti-symmetry in the field. So, for each symmetry seen in a geometry, one must carefully consider if several runs with different types of symmetry conditions are needed to reach the same total result as the full geometry would. This becomes especially important for some sub-physics such as structural mechanics or microacoustics, where shear forces come into play.

For structural mechanics there will typically be one degree-of-freedom per dimension, such that for a three-dimensional problem, one needs to resolve three displacements per point in the geometry, so any computational costs saved here will benefit the user. For standard pressure acoustics, there is only one degree-of-freedom to consider, namely pressure, but while there may be less saved in general by invoking symmetries, the acoustics domain are often rather large and could be the bottleneck in the setup. In general, one must consider which types of physics are involved, and which type of study is being run, before naively applying symmetrical conditions based on geometry only.

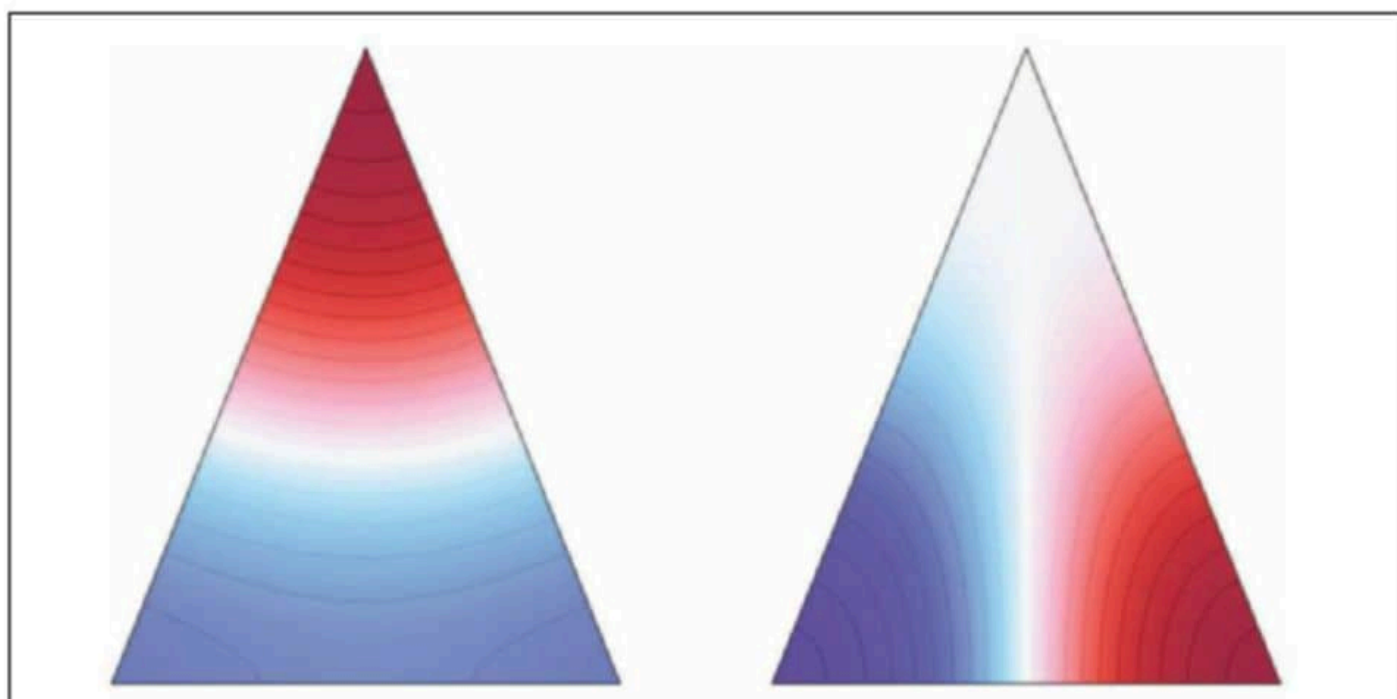


Figure 1: Two acoustic eigenmodes for a geometry with a single symmetry plane.

Different Conditions

Different conditions can sometimes be utilized in similar situations where a repeating pattern is present in the geometry, the so-called periodic conditions. These can be simple mirror conditions, anti-symmetry, or more involved such a Floquet conditions for repeating patterns on a grid. These periodic conditions may be applicable to such structures as those shown in **Figure 2**, but again this must be evaluated based on the physics, type of study, boundary conditions, and loads in question. Also note that if optimization is added on top of the simulation, additional considerations must be made, as the optimization may have additional symmetries [1].

Utilizing symmetry planes can reduce the overall computational cost by only considering a subset of an original geometry, but the dimensions of the remaining subset will be the same as the original. A different approach to reducing complexity in a simulation is what can be called “dimensional reduction,” where a three-dimensional domain is modeled using lower dimensions for some or all physics domains involved. This is commonly used in structural mechanics where “plane strain” or “plane stress” assumptions are made to reduce three-dimensional problems to two-dimensional ones, or in acoustics where for example a three-dimensional duct problem may be reduced to a one-dimensional one, under certain conditions. For transducers, where more than one physics is typically involved, several different dimensional reduction combinations can be constructed, and this will be illustrated here with a traditional moving coil loudspeaker problem used as the prototypical example.

One Common Dimensional Reduction Approach

One common dimensional reduction approach is to assume zero-dimensional behavior for electromagnetism and structural mechanics, with the effect of the acoustics environment included to some degree in a zero-dimensional fashion locally. This is done via lumped components, see for example [2], where the electromagnetic effects are included via one or more resistors and ditto inductors, and

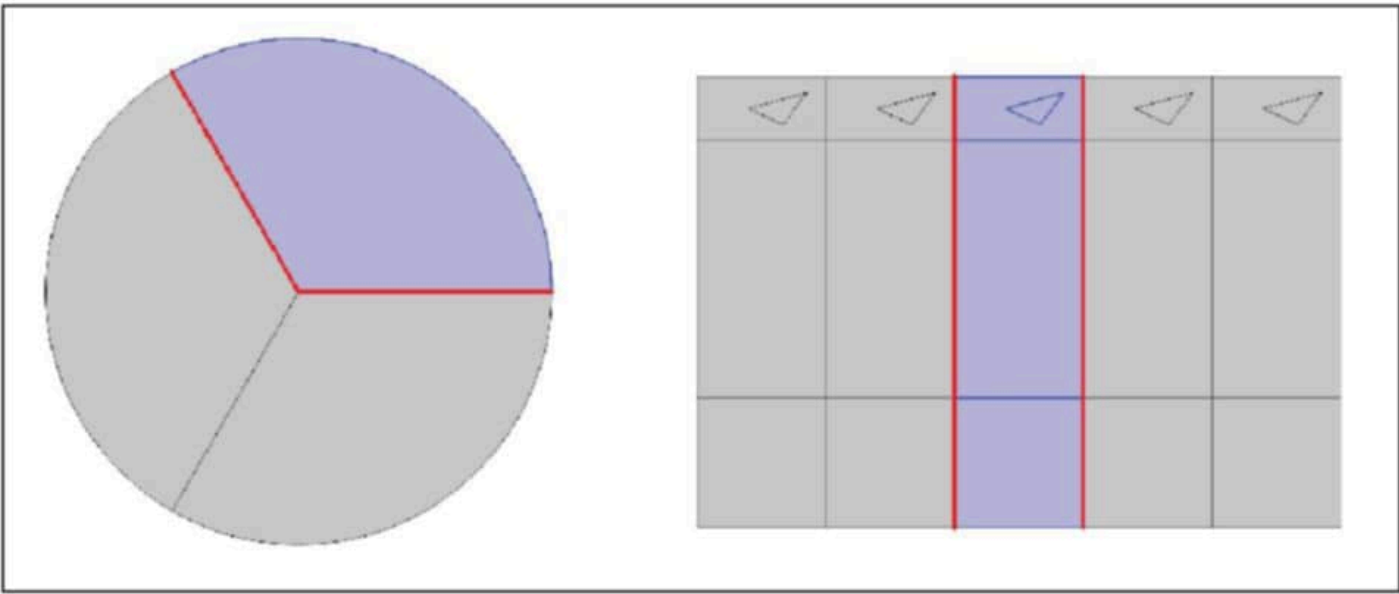


Figure 2: Geometries where periodic conditions should be considered on the red boundaries.

a total moving mass, a total stiffness, and a total loss are found for the loudspeaker driver, typically via measurements. The stiffness comes mainly from the spider and the surround, while the mass comes from all moving parts. While these parts each have their own mechanical characteristics, here they are lumped into single components, such that no modal behavior other than the fundamental for a single degree-of-freedom spring-mass system is included in the model. This is illustrated in **Figure 3**. The moving coil transductance principle is non-reciprocal and a coupling between the electromagnetics and structural mechanics is needed where the effort variable on the electromagnetic side voltage is coupled to the flow variable velocity on the structural mechanical side, and the flow variable current is coupled to the effort variable force, respectively. A so-called gyrator is utilized in the lumped circuit to obtain this coupling. Here, the zero-dimensional domain electromagnetics is coupled to the zero-dimensional domain via linking the mechanical velocity and the voltage u' in the circuit as:

$$Blv = u'$$

Additionally, the force applied to the vibroacoustics is linked to the current as:

$$F = Bli$$

The coupling between the mechanical and the acoustical domains is then implicitly included in that the impedance loading (typically denoted as “mass loading”) from the acoustical environment on to the structural mechanics domain is included via lumped components, such as capacitor for the compliance from a rear enclosure, and simple radiation conditions included via an impedance. Analogously,

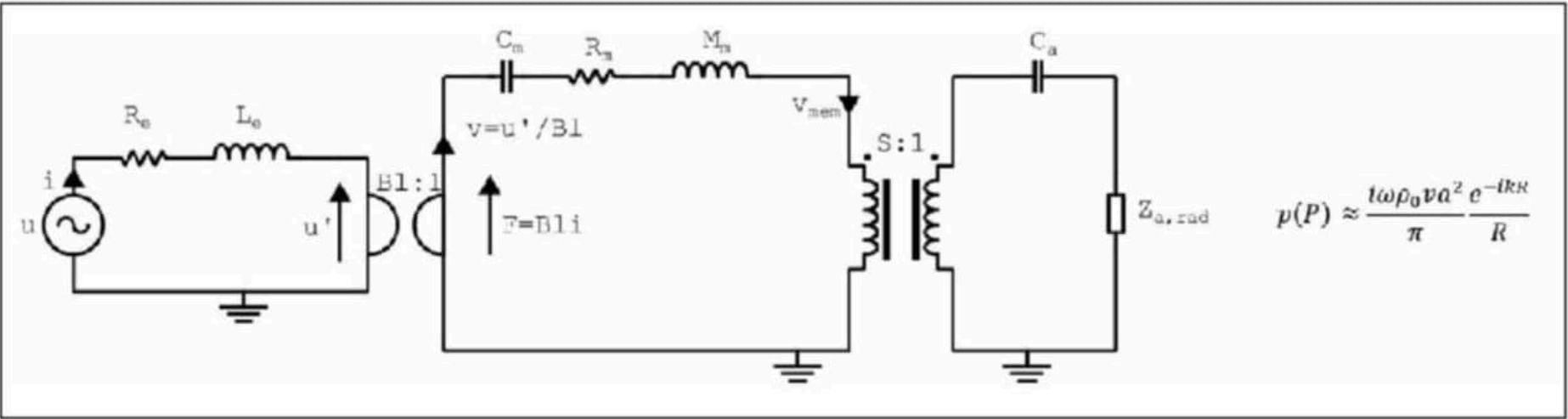
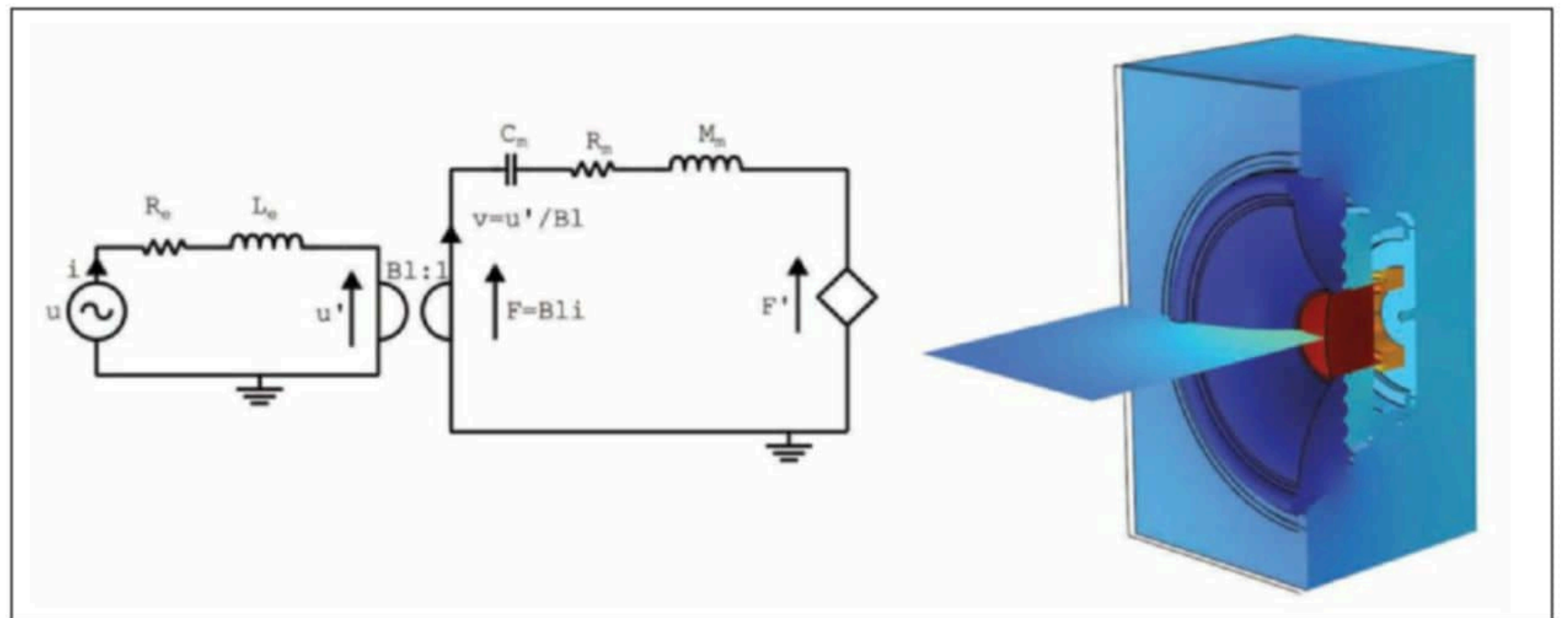


Figure 3: Standard lumped modeling with a [0D-0D-0D/Analytical] coupling.

Figure 4: A [0D-0D-3D] coupling resolving the front and rear acoustics explicitly.



the mechanical velocity found in the lumped circuit is assumed to be common for all parts, including the cone, and so this velocity is related to the acoustic excitation. However, the pressure of interest will almost always be at a certain distance away from the transducer and this pressure cannot be found locally in the lumped circuit.

This issue is typically dealt with via assuming the transducer acting as simple monopole source or a piston placed in a baffle and having an analytical expression for calculating the pressure at a distance. In the latter case, the Rayleigh integral is employed and simplified:

$$\rho(P) = \frac{i\omega\rho_0}{2\pi} \int_s v(Q) \frac{e^{-ikR}}{R} dS \approx \frac{i\omega\rho_0 va^2}{\pi} \frac{e^{-ikR}}{R}$$

The Rayleigh integral accurately gives the complex pressure for flat vibrating surfaces of arbitrary shape and arbitrary velocity distribution sitting in an infinite baffle, but in most loudspeaker literature this is further simplified by assuming that the surface is circular with radius a , has a constant velocity v across it, and that the pressure $p(P)$ of interest is on-axis and in the far-field. A two-way coupling is thus implicitly incorporated, as already described, but no modal behavior is incorporated for this piston approach, and no diffraction effects can be calculated with the acoustical components lumped in this manner.

More Complicated Acoustic Environments

For more complicated acoustic environments, to include diffraction for the enclosure and perhaps the inclusion of a complex rear chamber geometry, a different dimensional mapping can be utilized, where zero-dimensional components for the magnet system and structural mechanics part are used. These are then coupled to a three-dimensional acoustical domain, possibly with its own symmetry planes. The coupling takes place across the cone surface, and there are no moving parts in the finite elements analysis, although it may be advantageous to keep all explicit geometry parts to have the acoustics at the rear of the transducer better represented. This setup is shown in **Figure 4**, where the three-dimensional acoustic field is resolved with a coupling to the lumped mechanical components.

The lumped velocity is driving the acoustical velocity across the cone, surround, and dust cap, and here one needs to make choices about how this velocity is distributed and if the scalar velocity in the lumped model relates to a normal or an axial direction.

The coupling can be two-way in that not only does the zero-dimensional velocity drive the acoustical energy input, but also the pressure

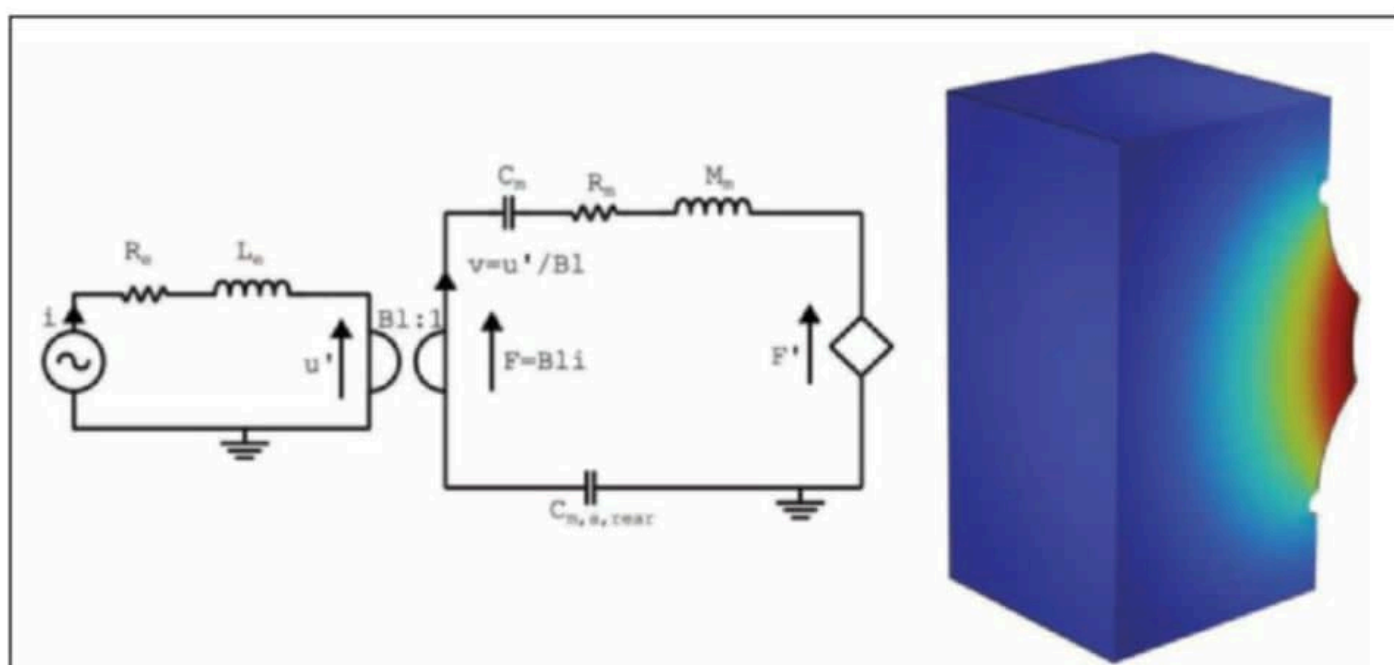


Figure 5: A [0D-0D-3D] coupling with lumped rear acoustics and explicit front acoustics.

About the Author

René Christensen (BSEE, MSc, PhD) has been working with simulations in the loudspeaker and hearing aid industry for several years, and in 2021 he started his own company, Acculution ApS, with a focus on mathematical modelling of many different products, such as transducers. The overall aim is to use academic research results for finding better engineering solutions faster, training engineers in this type of virtual prototyping, and working with students across the world to progress the field.



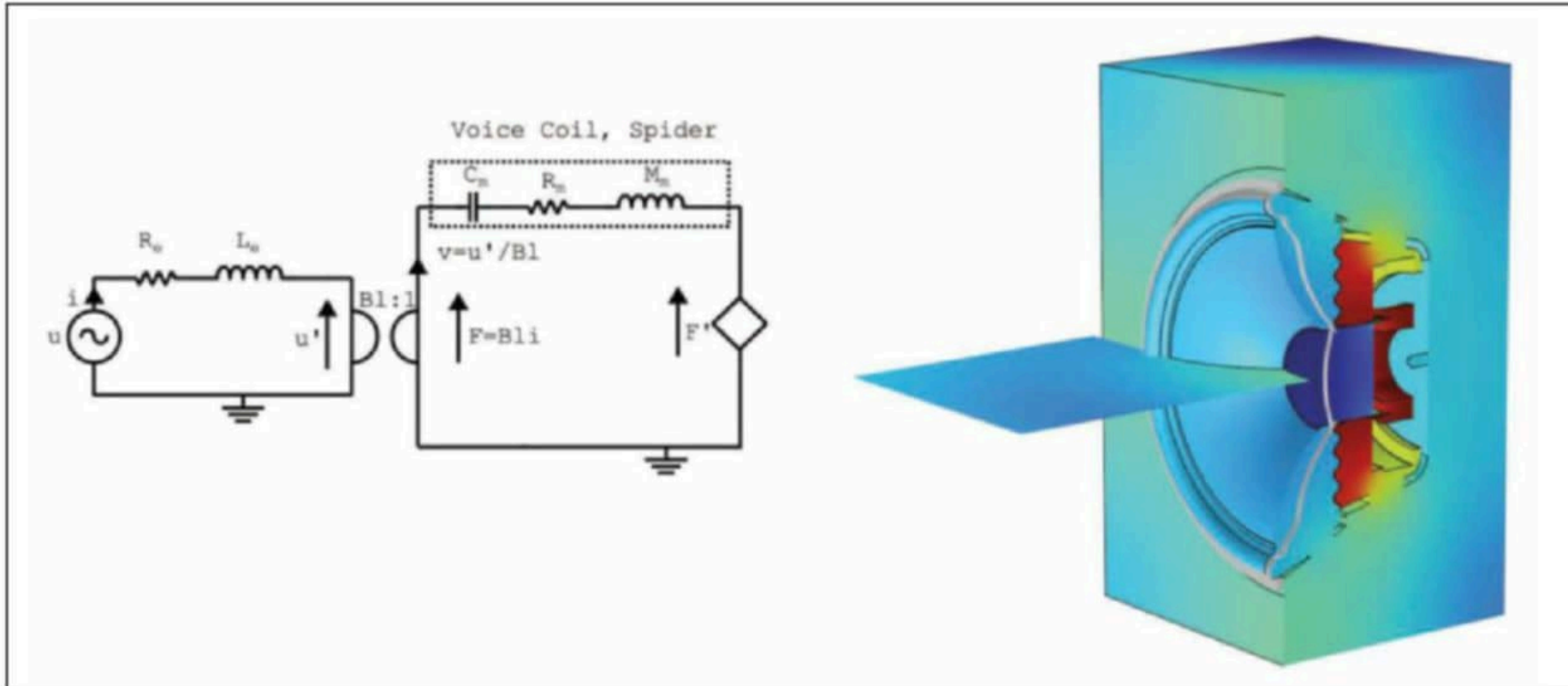


Figure 6: A [0D-0D-3D] mapping with lumped and explicit (grey) mechanical parts.

conditions across the cone can be coupled back to load the electro-mechanical system, so that:

$$F' = \int_s (\rho_{front} - \rho_{rear}) \vec{n} \cdot d\vec{s}$$

A simpler version of this coupling can be made for situations where the rear enclosure is included via a lumped component, and only the front acoustics is included, with or without its loading effect included. For some transducers, such as MEMS microtransducers, this approach will not be accurate enough as there can be intricate acoustic channels with microacoustical effects that need to be resolved, but for traditional loudspeakers, this may well be an approach worth pursuing.

The couplings for the moving coil transducers shown in **Figure 5** are sequential in that there is an interface between the electromagnetics and the structural mechanics, and between the structural mechanics and the acoustics. The dimensional splitting can be done directly at those interfaces, but there are alternatives such as splitting the dimension in between the acoustical parts, as just shown, but also in between mechanical parts. This was described in the literature [3] as a “hybrid” model around the time some of us were also doing it in the industry, and was years later described again [4], although the method was the same. The idea is to include the vibrating surface, typically cone, dust cap, and surround, as explicit domains, and lumping together the remaining moving parts on the rear, such as the spider and the voice coil (former plus windings), alongside a lumped magnet system. This setup is illustrated in **Figure 6**, where the cone, surround, and dust cap are now moving parts with structural mechanics physics applied, while the remaining mechanical parts are found in the

lumped circuit. While not emphasized in the literature, it has already been mentioned here that it may be important to have the lumped parts still included as geometry, as to better capture the rear acoustics, where the parts may form somewhat closed off regions with distinct resulting pressure characteristics. Also, as the former, windings, and spider have their own distinct mechanical modal behavior, these can influence the overall displacement from a structural mechanics perspective, and hence affect the sound pressure, and these effects are excluded in this model.

If all mechanical components are included explicitly, a setup is had, as illustrated in **Figure 7**, where only the electromagnetics are lumped, and the structural mechanics and acoustics are included in either 2D-axisymmetrically or three-dimensionally.

Here, the zero-dimensional electromagnetics domain is coupled to a higher-dimensional mechanical domain via linking the three-dimensional velocity and a zero-dimensional voltage as:

$$Blv_{avg,z} = u'$$

where an average velocity of the windings is calculated in the axial direction, here z-direction, as:

$$v_{avg,z} = \int_{\Omega_w} v_z(\mathbf{x}) d\Omega_w, \quad \mathbf{x} = (x, y, z)$$

There will typically be a two-way coupling, such that the pressures on the moving part couple with the normal stresses in the structural mechanics to have the acoustical loading included. This is a standard setup in most simulation software, but it will be computationally costly for many cases, since the acoustics environment will typically need three dimensions, and so the

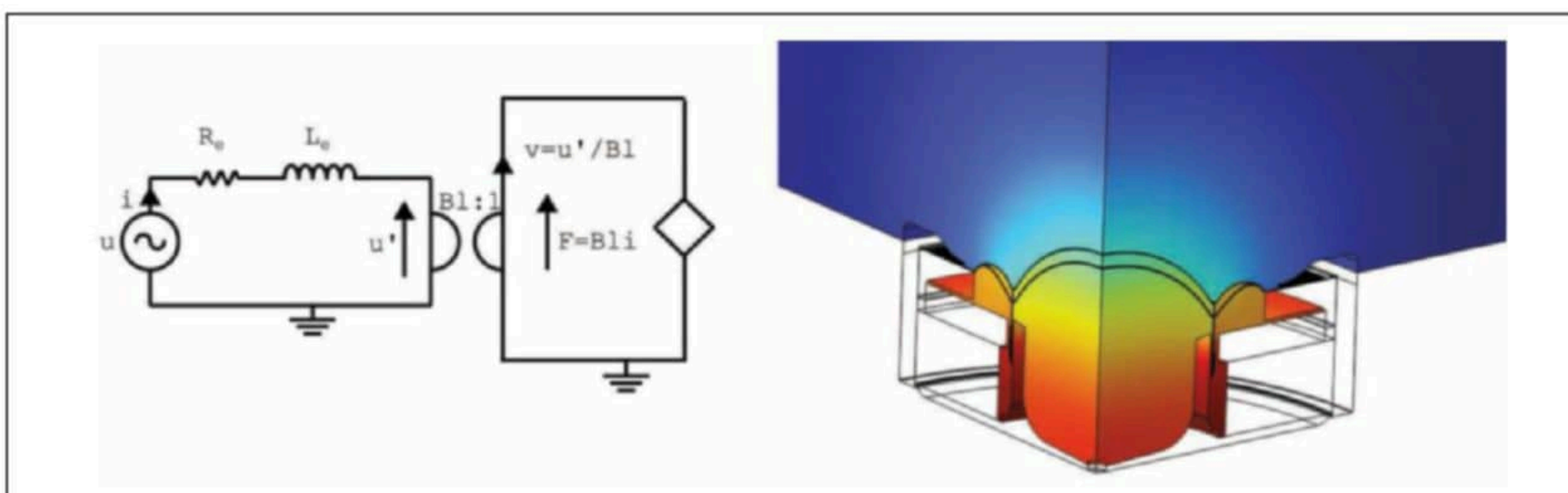


Figure 7: A [0D-2Daxi/3D] mapping of a tweeter with explicit mechanical parts in the 2Daxi or 3D domain.

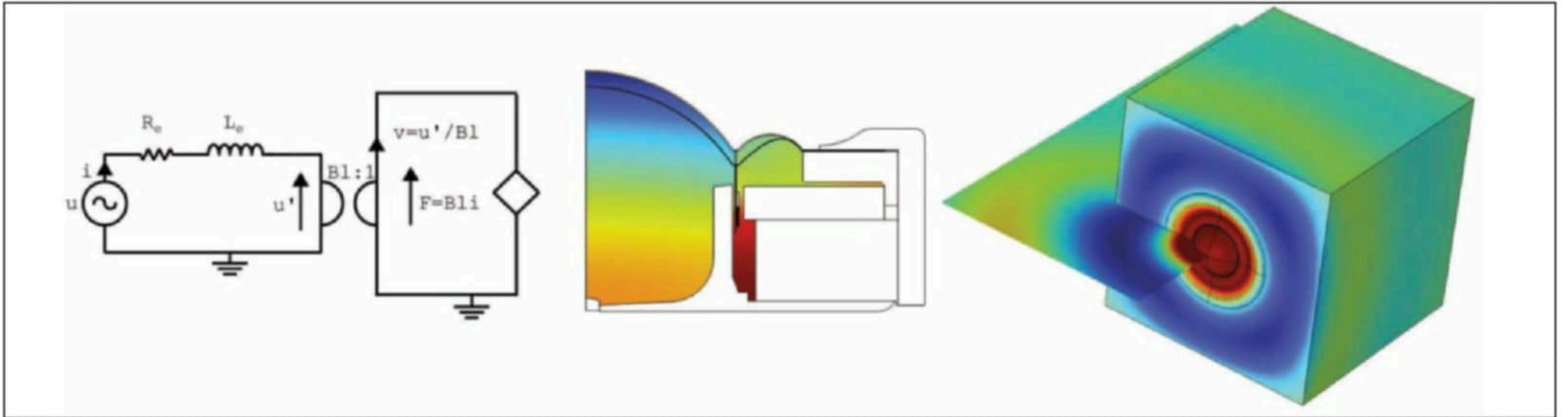


Figure 8: A [0D-2Daxi-3D] mapping of a tweeter with explicit mechanical parts in the 2Daxi domain.

mechanical counterparts also must be resolved in three dimensions, even if the transducer is inherently axisymmetric, at least in geometry, boundary conditions, and load.

Alternative Mapping

This leads to an alternative mapping that I have not seen described in the literature, shown in **Figure 8**. Here, the magnet system is included as lumped components, although it could be included in the 2D-axisymmetrical model for more elaborate magnet setups. The 2D-axisymmetrical domain includes all moving parts explicitly, and so modal behavior is included, albeit excluding circumferential variation. These variations may not be important as these modes are not necessarily excited much, and/or do not contribute much to the sound field, although this must be evaluated on a case-by-case basis. The rear acoustics geometry may be highly intricate, and can for example include microacoustics effects, porous materials, ferrofluids, or other characteristics that are not easily lumped.

Finally, the front acoustics geometry can be arbitrarily detailed, for example including enclosure diffraction, phase plugs, or living room or automotive environments. The transducer can thus be included in a very elaborate manner, while still saving substantial computational cost by only resolving it in 2D-axisymmetry, or 2D, depending on the transducer type, while the acoustic effects are captured near-perfectly. Different variations of this can be made, for example if the rear acoustics is simple enough to go into the lumped model, or alternatively too complex to be axisymmetric, such that the same approach as for the front acoustics is applied to the rear utilizing three dimensions for all acoustics domains.

This type of dimensional mapping is not trivial to set up, as the meridional velocity distribution on the transducer surface must be mapped to the corresponding meridional and circumferential points in the acoustics domain. It may be necessary to also have a two-way coupling incorporated. However, this overall strategy can take a simulation from requiring a high-performance cluster to solve, to solving locally on a more modest computer, and it is especially useful if additional studies such as topology optimization [5] are applied on top of it.

Conclusions

For all dimensional mapping strategies shown here, one must take care of the scalar and vector aspects of each sub-physics. Voltages and currents in a lumped model are complex scalars in zero dimension but they have signs relative to some arrow indications in the schematics. For any lumped model representing mechanical or acoustical parts, these signs relate to a particular direction, such as normal or axial, for underlying complex displacement vectors, or positive/negative aspects of scalar pressure fields.

Assuming standard acoustics, only normal displacements create sound, but if microacoustics effects are to be included, both normal and tangential displacements must be included. Also, single forces in a lumped circuit may be calculated from integral values coming from pressures in a three-dimensional domain. Getting all these signs and directions correctly implemented is crucial for the dimensional mapping to work.

For other transductance principles than the moving coil used as an example here, such as electrostatics or piezoelectric, the dimensional mapping will be different, and one needs to consider, for example, that they may or may not be reciprocal in nature. I am planning an article series on transductance principles, to go into more of these details.

References

- [1] R. Christensen, "Advancements in Acoustical Topology Optimization," COMSOL Conference 2023, Munich, 2023.
- [2] R. Christensen, "Lumped Element Modeling of Transducers," *audioXpress*, March 2023.
- [3] M. Pasi and M. J. Herring, "A Hybrid Electroacoustic Lumped and Finite Element Model for Modeling Loudspeaker Drivers," Audio Engineering Society 51st International Conference, Helsinki, Finland, 2013.
- [4] D. G. Nielsen, P. R. Andersen, J. S. Jensen and F. T. Agerkvist, "Estimation of Optimal Values for Lumped Elements in a Finite Element - Lumped," *Journal of Computational Acoustics*, Volume 28, No. 2, 2020.
- [5] R. Christensen, "Acoustic Topology Optimization for Vibroacoustics Applications," *audioXpress*, April 2023.

MENLO SCIENTIFIC

Engineering Consulting for the Audio Industry

We Know

- Product Development

We Provide

- Design and Technical Support

We Deliver

- Liaison, Sourcing, and Guidance

We've worked with the world's largest brands, supported large companies outsourcing production, advised manufacturing facilities on streamlining their process, and helped companies large and small to develop new products, and launch revolutionary technologies. We've helped startups find investors, and introduced entirely new concepts. Many companies learn their lesson by failing. Others have hired us first.

MENLO
SCIENTIFIC

Richmond, California, USA

- tel 510.758.9014
- mike@menloscientific.com
- www.menloscientific.com

40 Years



Mike Klasco
ALTI / ALMA Gold Driver
Award 2018



AES Life Member



ASA 25 Years



CES Innovation Awards
Judge 2024





Measuring MEMS Microspeakers

MEMS microspeakers are now commercially available and seeing fast evolution and adoption in true wireless stereo (TWS) earbuds and in-ear monitors (IEM). With the emergence of these small form-factor transducers comes the need to address how to measure its performance and ensure product quality.

By

Steve Temme (Listen Inc.) and **Michael Ricci** (xMEMS)



MEMS microspeakers are driving one of the fastest technological transitions in the miniature speaker market. MEMS transducers first gained widespread attention in the late 2000s with the first commercially available MEMS microphone, and over the course of a decade, they replaced electret microphones in the majority of mobile devices. The MEMS speaker revolution is shaping up for an even faster transition. Early speakers incorporating MEMS technology were launched in 2014, but it was not until 2021 that the first fully piezoelectric MEMS microspeakers became commercially available. Manufacturers are already rapidly adopting these into TWS and pro-audio IEMs, earning praise from consumers and product reviewers alike, and propelling this technology into the mainstream.

MEMS Speaker Overview

MEMS microspeakers are constructed in an entirely different way from conventional miniature speakers. Rather than using inductive coils and magnets, they rely on a voltage-driven capacitive actuator, and they are entirely manufactured using a monolithic solid-state fabrication process in a semiconductor wafer foundry. This fabrication process delivers much tighter speaker-to-speaker consistency than is possible with a conventional driver made up of an assembly of many moving parts. They offer full-range performance, eliminating crossover networks that introduce complexity and sometimes even phase differences. This is particularly important considering this year's other big audio trend—spatial audio, where the phase response is critical to accurate reproduction. Piezo-MEMS speakers are also compact in size and low power, making them economical on both space and power consumption. Finally, they are surface-mount technology (SMT) reflowable, offering yield and quality advantages in device manufacturing.

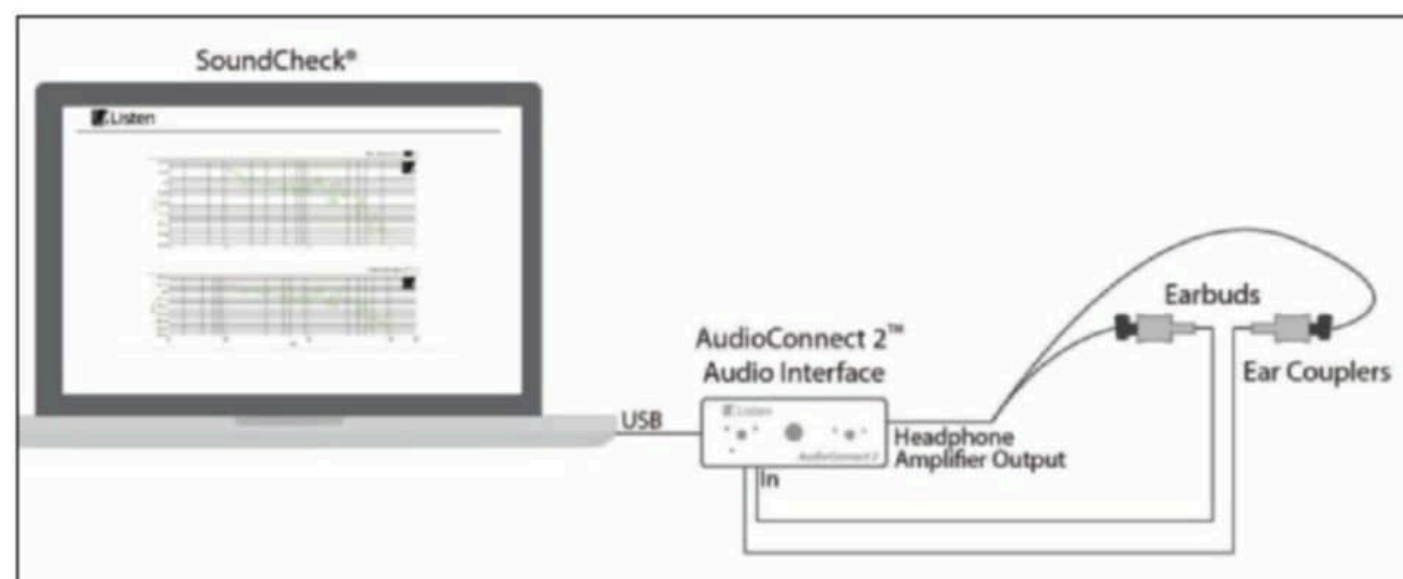


Figure 1: Test configuration for a MEMS earbud measurement.

While these advantages are leading to rapid adoption in devices such as hearing aids, earbuds, professional in-ear monitors, smart glasses, and more, they are not without their challenges. The maximum sound pressure level (SPL) in the low frequencies is typically less than is required for a leak-tolerant active noise cancelling (ANC) system compared with conventional headphone drivers, limiting their use for ANC earbuds to a mid/high frequency tweeter in a two-way configuration. However, in recent months, manufacturers have made rapid progress in overcoming these limitations, thus broadening their applications. For example, xMEMS' new Cypress Piezo-MEMS speaker, showcased at the recent Consumer Electronics Show (CES), delivers up to 140dB SPL in the low-frequency band—more than sufficient for ANC applications. The cost of MEMS-driven products is also decreasing as this technology is more widely adopted. While early MEMS earbuds were priced well over \$1000, recently launched models such as the Creative Labs Aurvana bring this technology's high-quality sound to TWS products below a \$200 price point. As this technology moves toward becoming the de facto standard, the outlook for this market is positive.

The Importance of Measurements

Any time a new technology is introduced, measurements are critical. Designers rely heavily on models to predict performance before building devices. While such models are well established and based on decades of data for conventional drivers, models for MEMS speakers are still in their infancy. It is therefore critical to validate and iterate models with actual measurements for them to be valuable design tools. Measurement is also critical for both MEMS device manufacturers and those companies using their products to thoroughly evaluate performance and ensure product quality.

Measuring Finished MEMS Headphones

Finished products such as earbuds are simple to measure. A MEMS headphone can be tested in exactly the same way as any other headphone using an audio test system with a head-and-torso simulator (HATS) or ear couplers, and a Bluetooth interface if needed. **Figure 1** shows a typical headphone test configuration that could be used for testing a pair of wired MEMS earbuds. The test method is simple. The test signal is played through the headphones and captured using a HATS or ear couplers, where it is sent to the computer for detailed analysis of frequency response, phase, distortion, and more.

Measuring MEMS Drivers

Raw MEMS drivers are more complicated to measure due to their small form-factor and power requirements. That said, the basic measurement principles are the same as any other driver; the complexity is in the fixturing. In addition to an audio test system (interface plus software), a typical configuration requires an ear simulator coupler to capture the signal, custom fixturing to hold the MEMS driver in place, and a special driver circuit for the MEMS device. Piezo-MEMS devices have a very high input impedance and draw very low current. Typically, MEMS speaker manufacturers provide their own driver circuit to deliver voltage bias and boost converter to step up the voltage, and this should be used for making measurements.

Figure 2 shows a schematic diagram of the test configuration used to measure a MEMS speaker, and **Photo 1** shows the test jig, coupler, and charge amplifier in more detail.

To measure the Piezo-MEMS speaker, an xMEMS Montara Plus (**Photo 2**), the xMEMS “Aptos” Class H charge amplifier supplies the required bias voltage and up to 24dB of gain to power the device. A 3D printed device test jig is connected to a 711 ear simulator coupler to simulate an in-ear measurement. An AudioConnect 2 audio interface powers the coupler and is connected to SoundCheck for analysis. Aside from the charge amplifier and custom fixture, the test setup is very similar to that used for a conventional speaker.

The test itself is a standard speaker test, with levels and frequency range set appropriately for the device. It plays the stimulus (e.g., a stepped sine wave) and analyzes the response, displaying frequency response, impedance, and distortion.

MEMS Measurement Results

The frequency response, shown in **Figure 3**, is very flat up to 1kHz. In the higher frequencies, there are a couple of resonances—one due to the

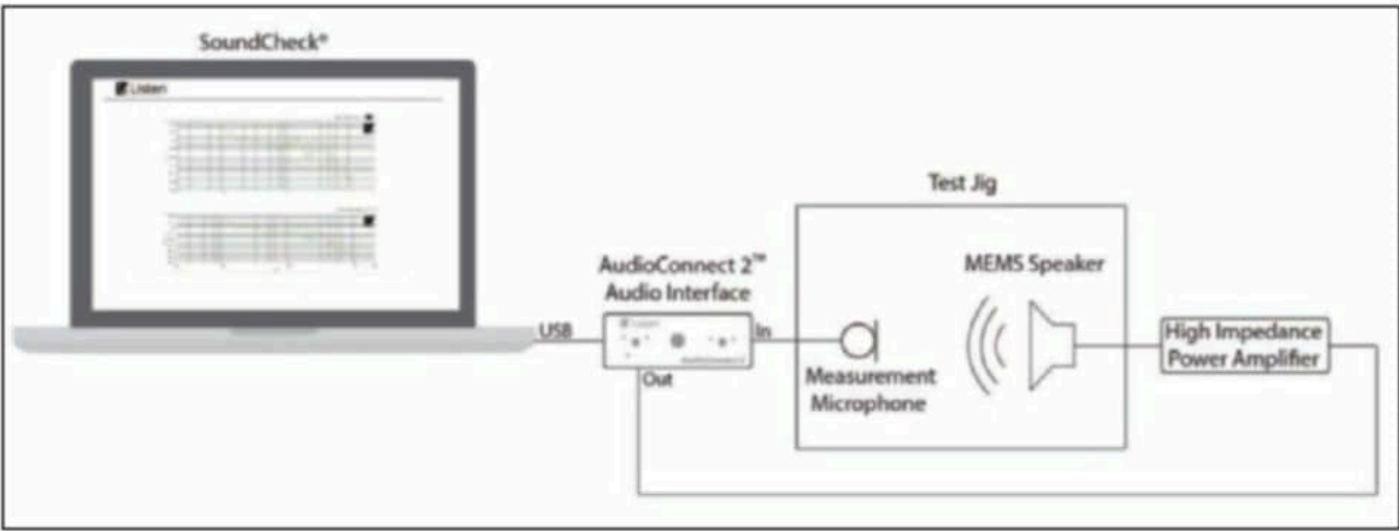


Figure 2: Test configuration for MEMS driver measurement.

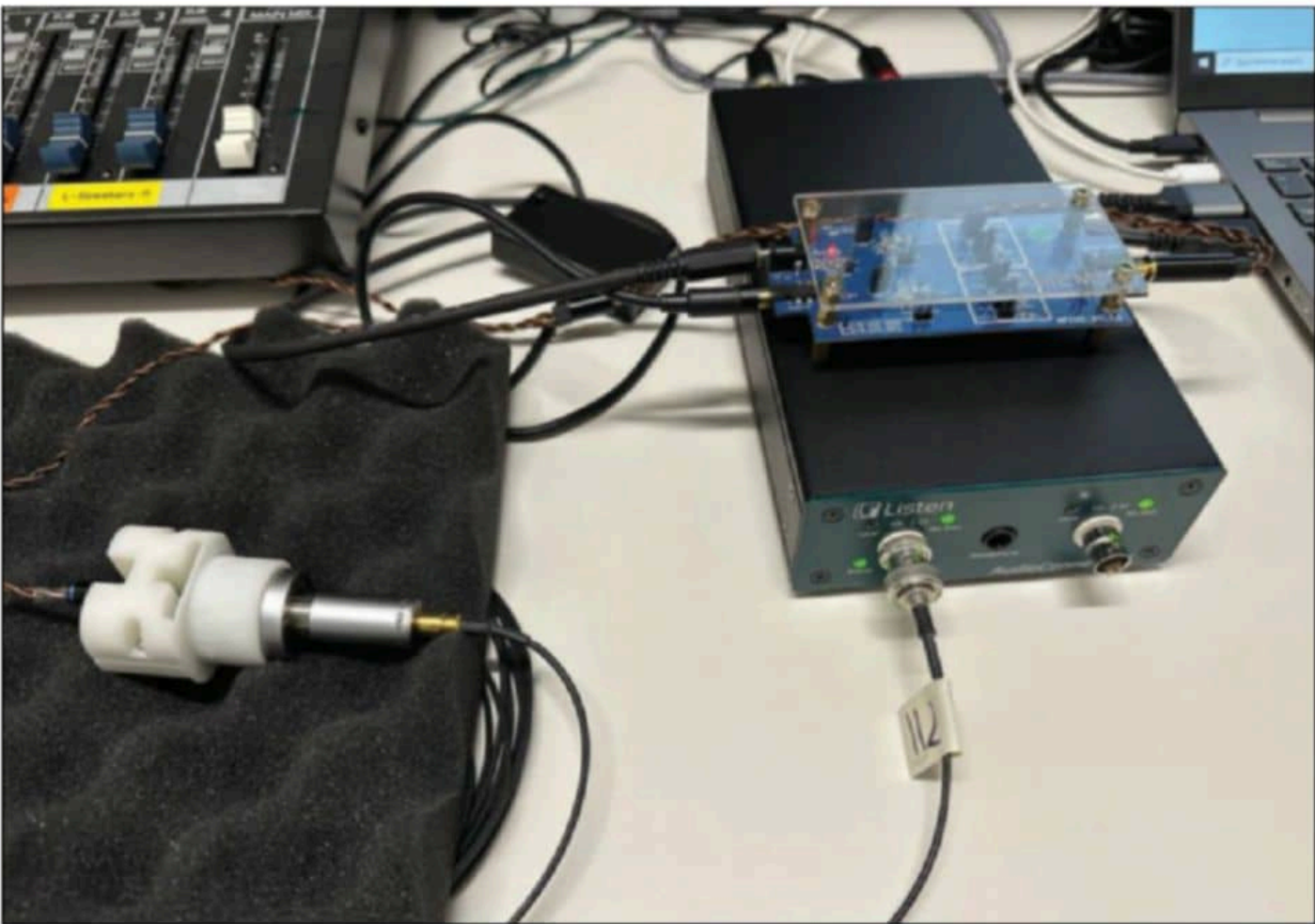


Photo 1: Charge amplifier and test fixturing for a MEMS earbud measurement.

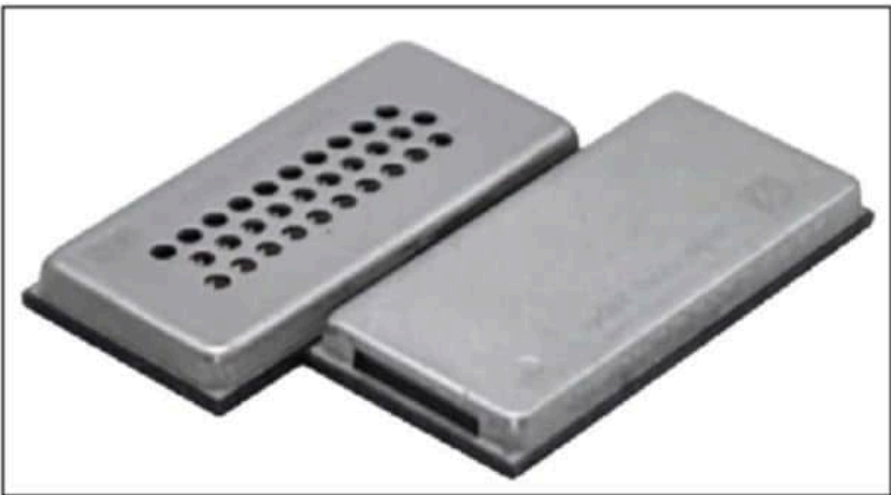


Photo 2: Montara Plus MEMS speaker.

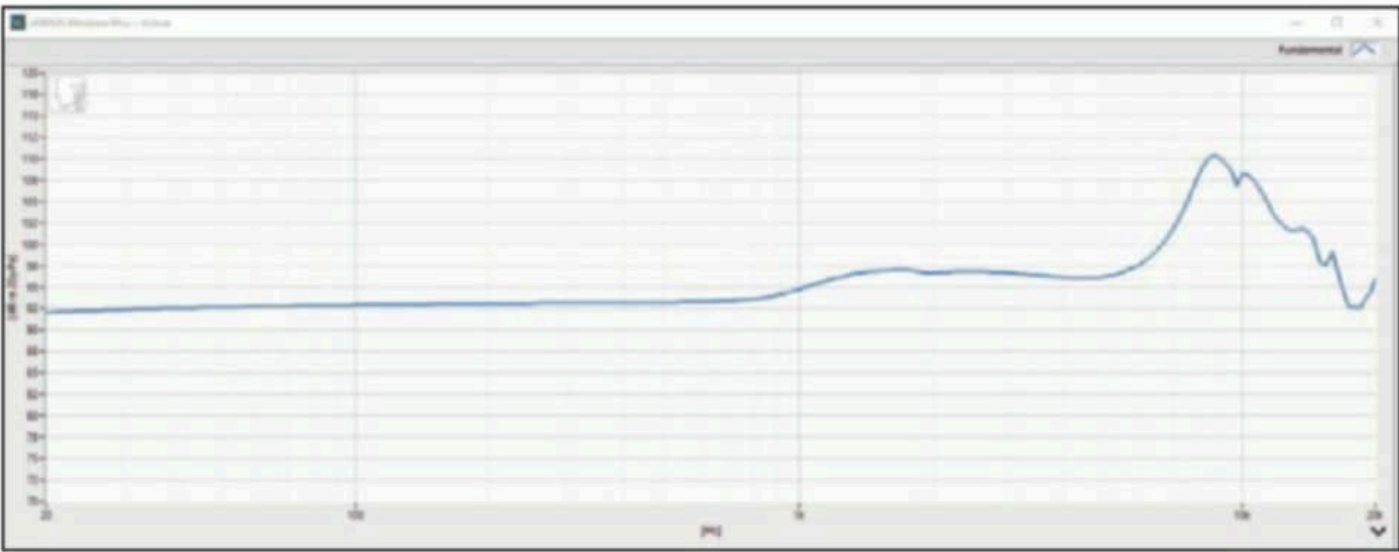


Figure 3: Frequency response of Montara Plus MEMS speaker.

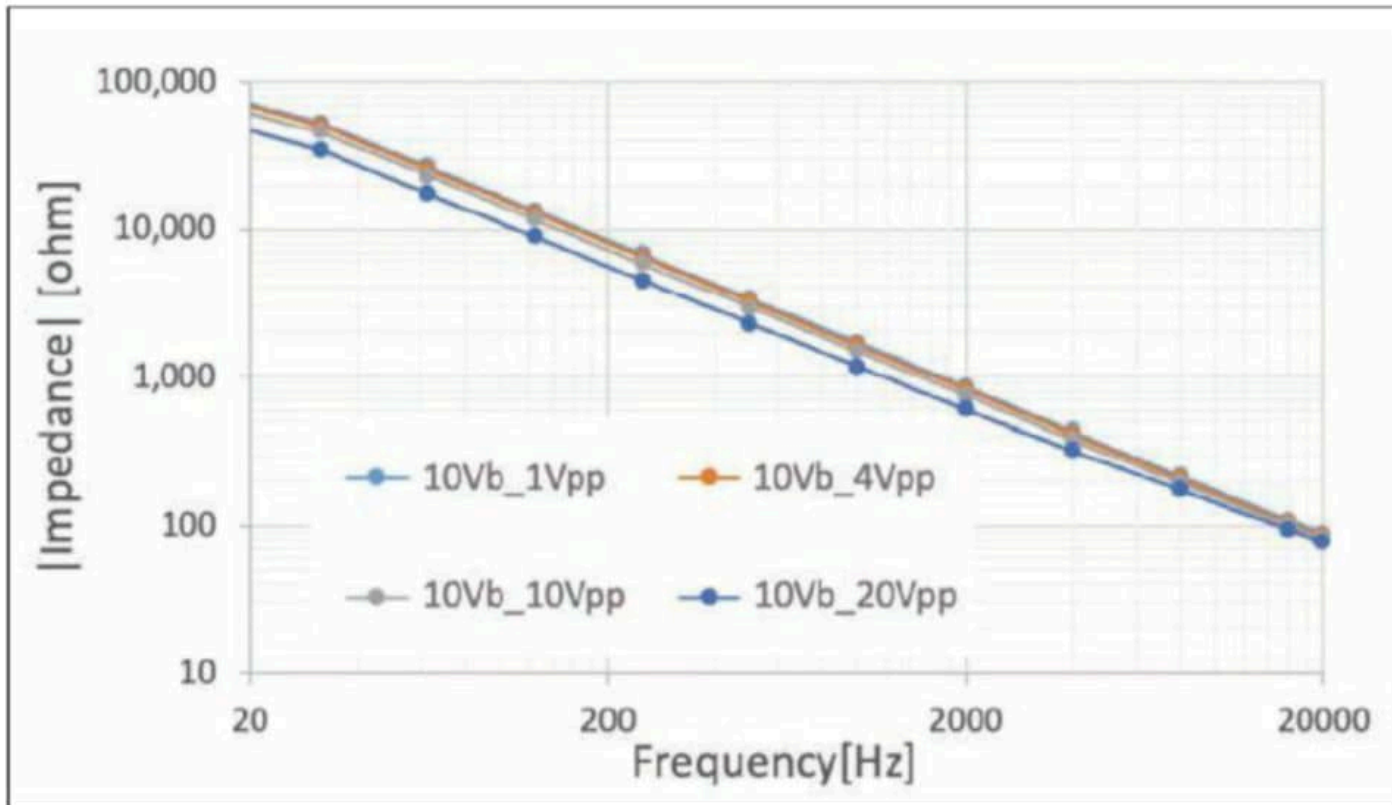


Figure 4: Typical impedance curve for a Piezo-MEMS microspeaker.

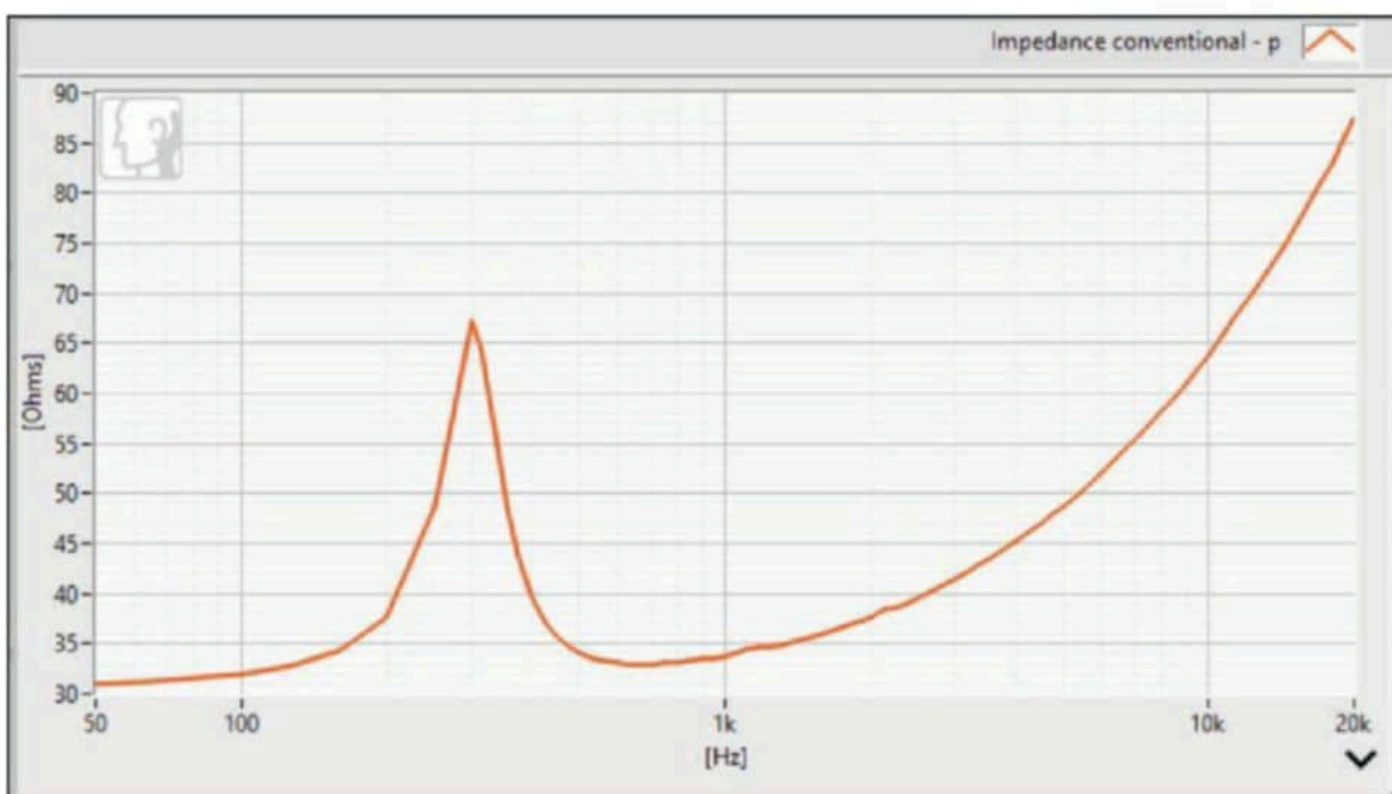


Figure 5: Impedance curve of a small conventional speaker (shape for comparison).

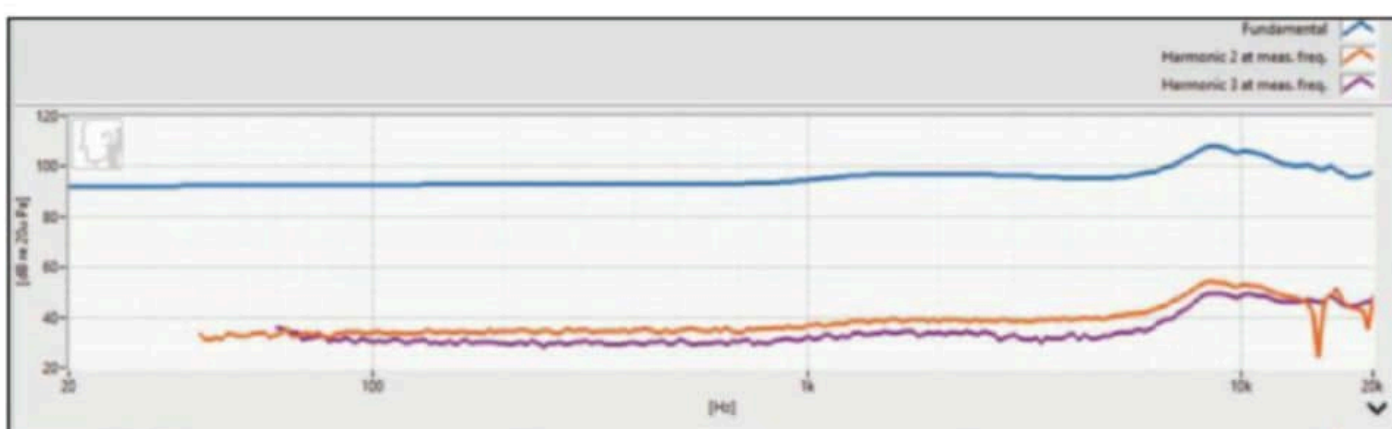


Figure 6: Second and third harmonics of a Piezo-MEMS microspeaker plotted at their measured frequencies.

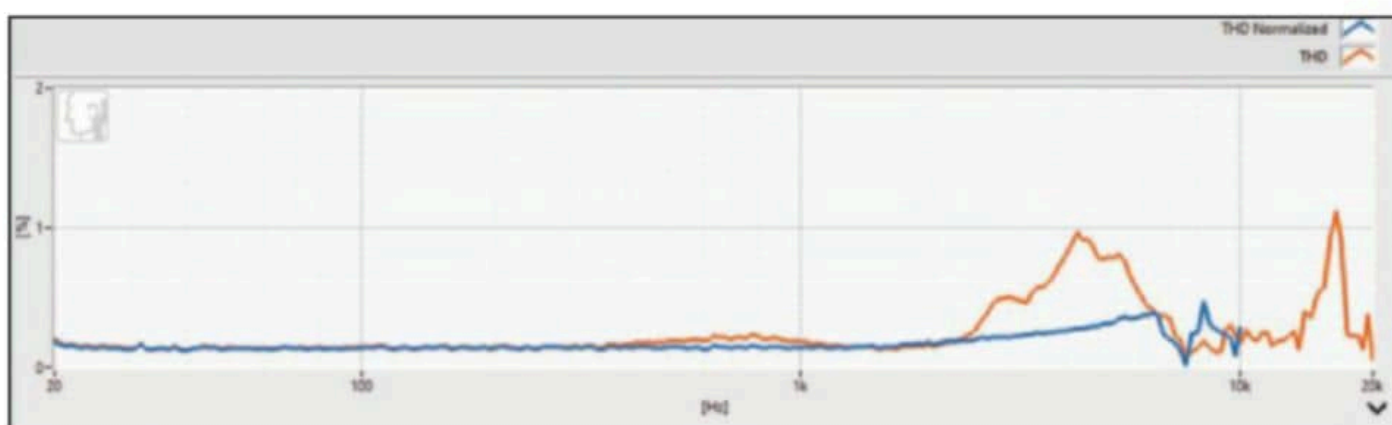


Figure 7: Frequency normalized THD (blue curve) compared to traditional THD (orange curve) for a Piezo-MEMS microspeaker.

piezoelectric material and the other due to the resonance of the coupler, which simulates the human ear. MEMS speakers exhibit fewer resonances than conventional speakers due to the simplicity of their design. They have fewer moving parts than a conventional speaker and consequently fewer break-up modes. Once the device is measured and the resonances known, they can be accounted for in the product design to minimize their effect. The higher frequency resonances in the MEMS speakers are actually easier to design around as they are out of the most sensitive range of the human ear. Perhaps future development might result in these resonances being moved out of the audible range altogether. Naturally this information also provides useful data points for refining MEMS speaker models, which will, in turn, help design better speakers.

Figure 4 shows the impedance of the MEMS speaker. You can see here that it's a very different shape from a conventional loudspeaker impedance (**Figure 5**). Although the values are much higher, it's very linear, which makes it easy to compensate for when designing products.

It's advantageous to use frequency-normalized distortion measurements rather than conventional distortion measurements when measuring the total harmonic distortion (THD) of MEMS speakers or any other loudspeaker, as the difference can be quite significant. Normalized distortion measurements compare the harmonic levels to the fundamental level at their measured frequency before their ratio is plotted, rather than the fundamental level at the excitation frequency. This provides more accurate results as it removes the effect of the non-flat frequency response from the THD. This makes it easier to see the peaks in the THD response independent of the peaks and dips in the fundamental response, and easier to set limits.

Figure 6 shows harmonics plotted at their measured frequency (not at their excitation frequency) along with the fundamental for the Montara Plus piezo-MEMS speaker. You can see that the fundamental bumps at the high frequencies are superimposed on the harmonics.

In **Figure 7**, the orange line shows the THD calculated at the conventional stimulus frequencies (THD) and the blue line shows the normalized THD, calculated at the measured stimulus frequencies. Removing the effect of the non-flat frequency response from the distortion makes it easier to see the peaks in the distortion response independently of the peaks and dips in the fundamental response.

It shows that the THD is very low up to 10kHz. As with the frequency response, an understanding of the high-frequency distortion enables it to be

minimized with good product design. It is clear from the normalized measurement that there is a high Q at resonance between 8kHz and 9kHz, but apart from that there is very little distortion, so design efforts can be focused on planning around that one peak. If a design was based on the conventional distortion measurements shown by the orange curve, time would be wasted trying to solve the distortion between 4kHz and 5kHz on the graph that does not really exist.

Conclusions

MEMS speaker technology is rapidly evolving to meet today’s mobile audio requirements. The simplicity of construction results in features such as flat frequency response, linear impedance, and predictable distortion that facilitates their design into consumer electronics products.

Accurate audio measurements of both raw drivers and finished goods are essential to develop and refine models, which are still in their infancy compared to more established driver technologies. While finished goods containing MEMS devices can be measured in the same way as their conventional counterparts, raw MEMS drivers require custom fixturing and an appropriate amplifier. With these test components, they can be measured in the same way as a conventional driver.

The distortion in a MEMS speaker is typically highly focused around one frequency, and normalized THD, often called frequency normalized distortion, is highly recommended over conventional or non-normalized distortion measurements to accurately evaluate this distortion. 

About the Authors

Steve Temme is founder and President of Listen, Inc., manufacturer of the SoundCheck audio test system. Steve founded the company in 1995, and over the past 29 years the company has remained on the cutting edge of research in audio measurement, introducing many new measurement algorithms and techniques. In recent years, Listen’s product development focus has been on creating the algorithms, interfaces and test methods to measure today’s newest devices including MEMS speakers and microphones, as well as automotive audio and various wireless and cloud-communicating devices. Prior to founding Listen, Steve worked for many years as an acoustic test and measurement applications engineer at Brüel & Kjær, and also as a loudspeaker design engineer at Apogee Acoustics. He holds a BSME from Tufts University, has authored numerous papers on acoustic testing, and has lectured extensively throughout the world.



Michael Ricci is the Senior Director of Electroacoustic Engineering at xMEMS Labs, in Santa Clara, CA, where he built and maintains multiple test labs to support Piezo-MEMS μ Speaker integrations for consumer hearables and professional audio products. His technical background includes 25 years of semiconductor device testing with a focus on MEMs transducers. Michael has held Senior FAE and Principal Electroacoustic Engineering positions with Bosch Sensortec (Akustica MEMs Mics) and Knowles Electronics supporting strategic accounts with MEMs microphone and balanced armature integrations. Industry affiliations include Acoustical Society TAG voting member, the Audio Engineering Society (AES), the National Association of Music Merchants (NAMM), and the American Society of Composers, Authors, and Publishers (ASCAP).





WARKWYN
Consultation • Analysis • Sales

Warkwyn utilizes and represents the most sophisticated analysis and measurement equipment available.



#1 Electro-Acoustic Hardware and Software Measurement Systems in the World.



Incredibly Unique Particle Velocity Measurement Systems for Noise and Vibration Localization. Hear it **and** See it.

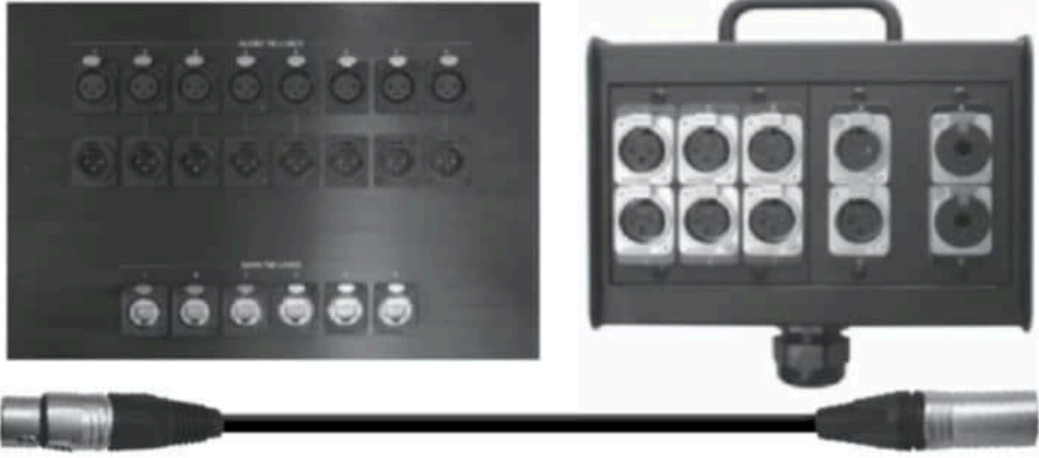


Precision Made Type 1 & 2 SPL Measurement and Vibration Analysis Systems, Committed to a Quieter World Since 1969

For more information on Warkwyn’s services, or for more information on our fantastic partners, please contact info@warkwyn.com or 612.759.4042 www.warkwyn.com

DO IT YOUR WAY
Panels/Boxes/Cables
WITH REDCO AUDIO’S
“Build Your Own” web pages

Full control for design, editing and more.
WE’LL BUILD IT JUST THE WAY YOU WANT!

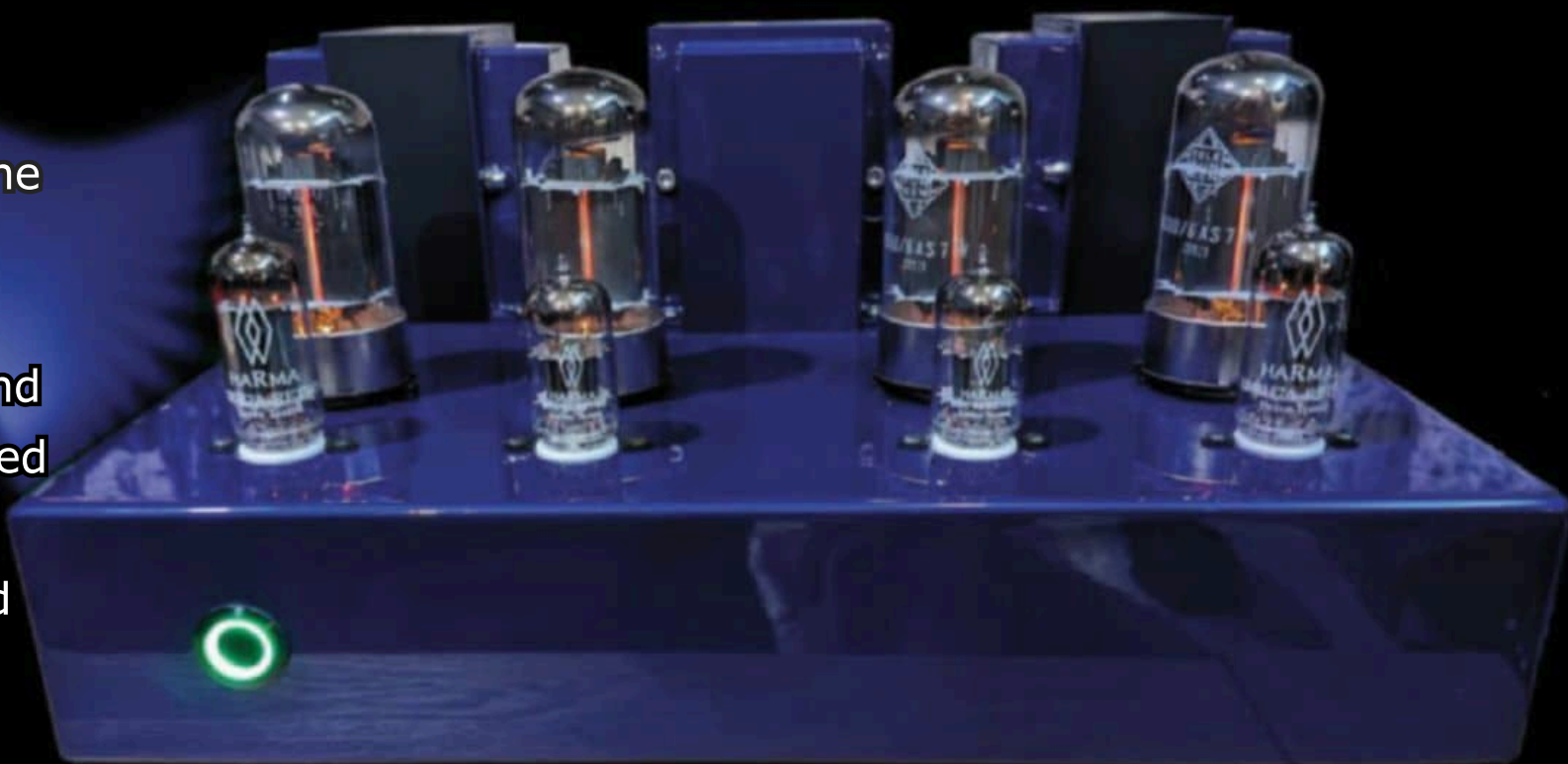


REDCO AUDIO
www.redco.com  **203-502-7600**

A Low-Feedback Triode Amplifier with Zero Output Impedance

Part 2 – Balanced Input Amplifier Design, Performance, and Detailed Calculations

The first part of this article introduced the Zero Output Impedance (ZOI) amplifier concept and discussed the design for a single-ended input version. In this second part, the design is extended to a balanced input configuration. Also, performance measurement figures are discussed, and two Appendices give detailed design calculations for both configurations.



By
Tim Mellow and Peter Farrow

The scheme for a balanced input is shown in **Figure 5**. The input signal voltage is represented by V_i and the output voltage across the loudspeaker of impedance R_L is represented by V_o . For simplicity, the amplifier is represented by an op-amp feeding a transformer of arbitrary turns ratio. The important thing is that the gain between the differential input and the secondary winding is A , which is the open-loop no-load gain of the amplifier. By open loop, we mean the gain with the feedback resistors R_2 and R_3 removed. The open-loop output impedance of the amplifier is represented by R_0 . A small-value current-sensing resistor R_4 is connected in series with the loudspeaker so that current feedback is routed to

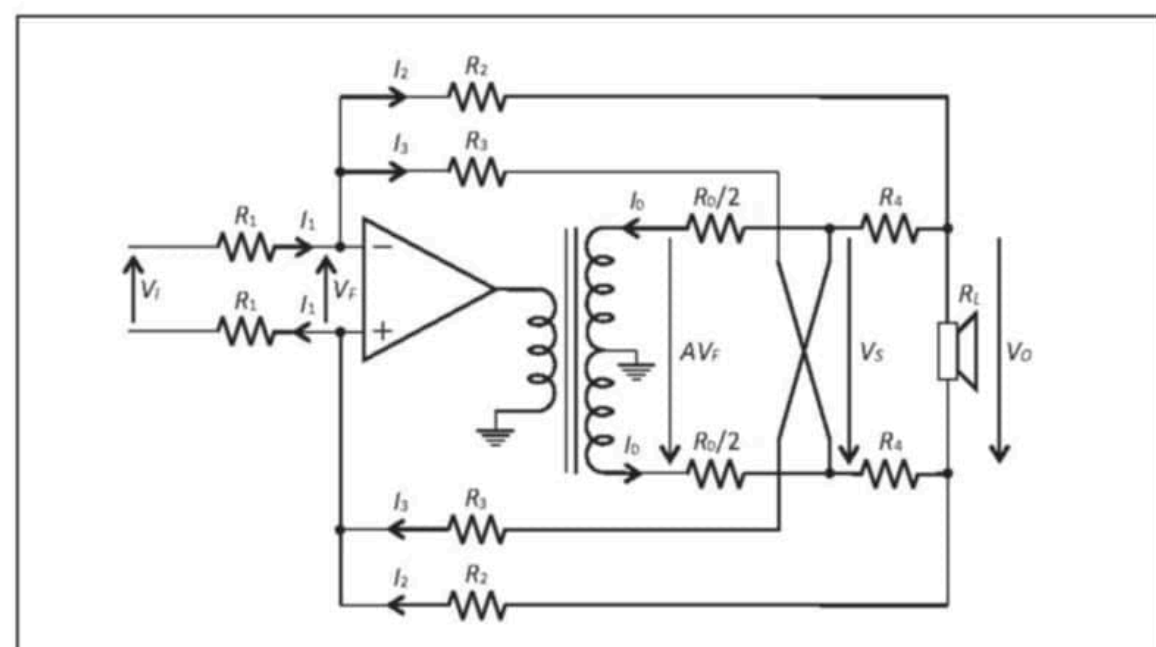


Figure 5: Outline schematic of the ZOI scheme for a balanced input as devised by the current author.

the input via R_3 and voltage feedback is routed to same via R_2 . The feedback voltage V_F is developed across the non-inverting and inverting inputs. A complete schematic for a triode amplifier using the ZOI scheme for a balanced input is shown in **Figure 6**. The associated power supply is shown in **Figure 7**. The Parts List details the items needed to build the triode amplifier (**Photo 1**), using the ZOI scheme for a balanced input.

A full derivation in Appendix 2 leads to the following formulas for R_2 and R_3 :

$$R_3 = \frac{2A((A-1)R_4 - R_0)R_1R_L}{(F(R_0 + 2R_4 + R_L) + AR_L)(R_0 + 2R_4)}$$

$$R_2 = \frac{(A+1)R_1R_3R_L}{F(R_0 + 2R_4 + R_L)R_3 + ((A-1)R_1 - R_3)R_L}$$

In the upper half of Figure 6, we have kept the same resistor numbers R_1 , R_2 , R_3 , and R_4 as in Figure 5 so that they can be easily identified. However, to avoid duplicating component designations, we have used R_{21} , R_{22} , R_{23} , and R_{24} , respectively in the lower half. In this case, we measure an open-loop gain $A = 55$ and output impedance $R_0 = 2.4\Omega$. We assume a load impedance of $R_L = 6\Omega$ and set the feedback factor to $F = 2$ (6dB). If we set $R_1 = 22\text{ k}\Omega$ and $R_4 = 0.1\Omega$, then the above formulas give $R_3 = 48.3\text{ k}\Omega$ and $R_2 = 46.5\text{ k}\Omega$. The values shown in Figure 6 are the nearest preferred values.

Apart from the feedback connection, the balanced circuit of Figure 6 is similar to the single-ended one of Figure 3 (see

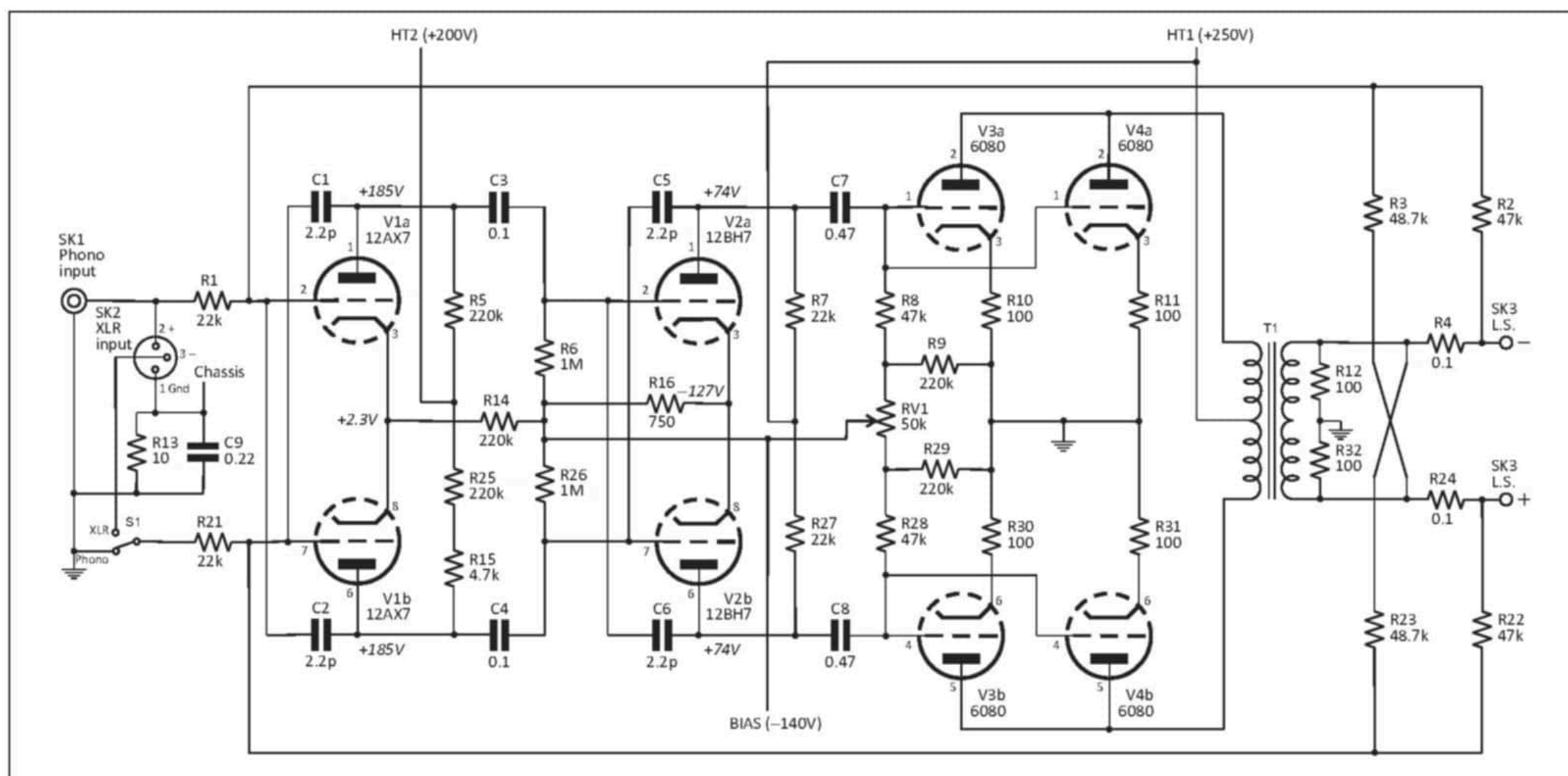


Figure 6: Schematic of a triode amplifier using the ZOI scheme for a balanced input.

Part 1, *audioXpress* May 2024). However, the connection from the anode of V1a to the grid of V1b has gone and is replaced entirely by cathode coupling. Hence, the cathode resistor R14 is much larger. To facilitate this, the grids, as well as the ground, are referenced to the positive side of the bias supply, not the negative. Hence, the cathode resistor R14 has the entire 140V of the bias supply across it, which sets the total current through V1a and V1b to 0.64mA. To ensure that their outputs have equal magnitude, a small-value resistor R15 is added to the anode load of V1b.

As well as having a balanced input, the amplifier needs a balanced output to ensure correct operation of the feedback. One could use a center-tapped winding on the output transformer with the center tap connected to ground but the routing out of the thick secondary wire would reduce the space available for the remaining windings and thus reduce efficiency. Hence, we use resistors R12 and R32 connected to ground instead.

Photo 2 shows a practical implementation. The channel on the left is wired for a balanced input while that on the right is wired for a single-ended input.

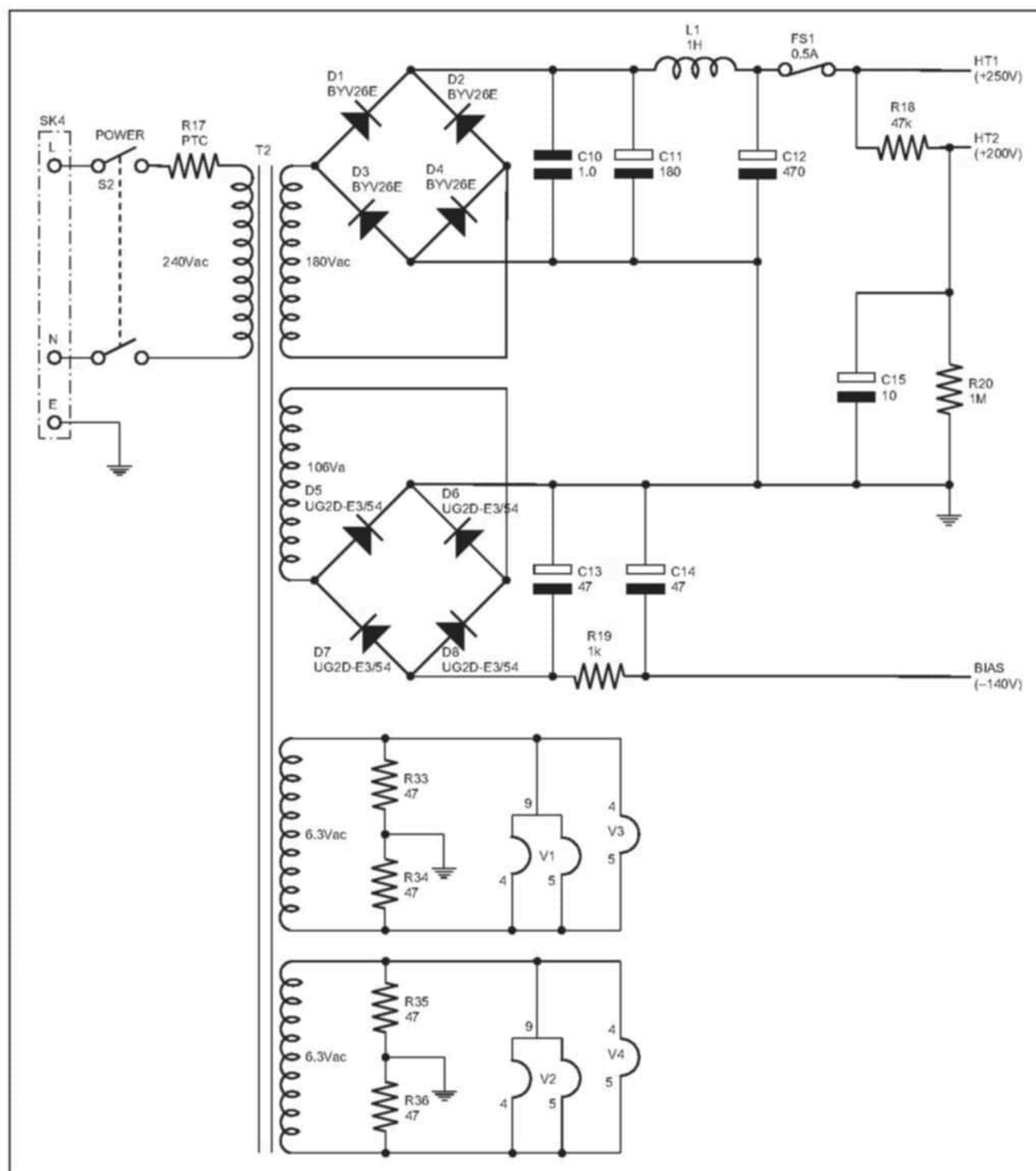


Figure 7: Schematic of a power supply for triode amplifier using the ZOI scheme for a balanced input.



Photo 1: This is the triode amplifier using the ZOI scheme for a balanced input.

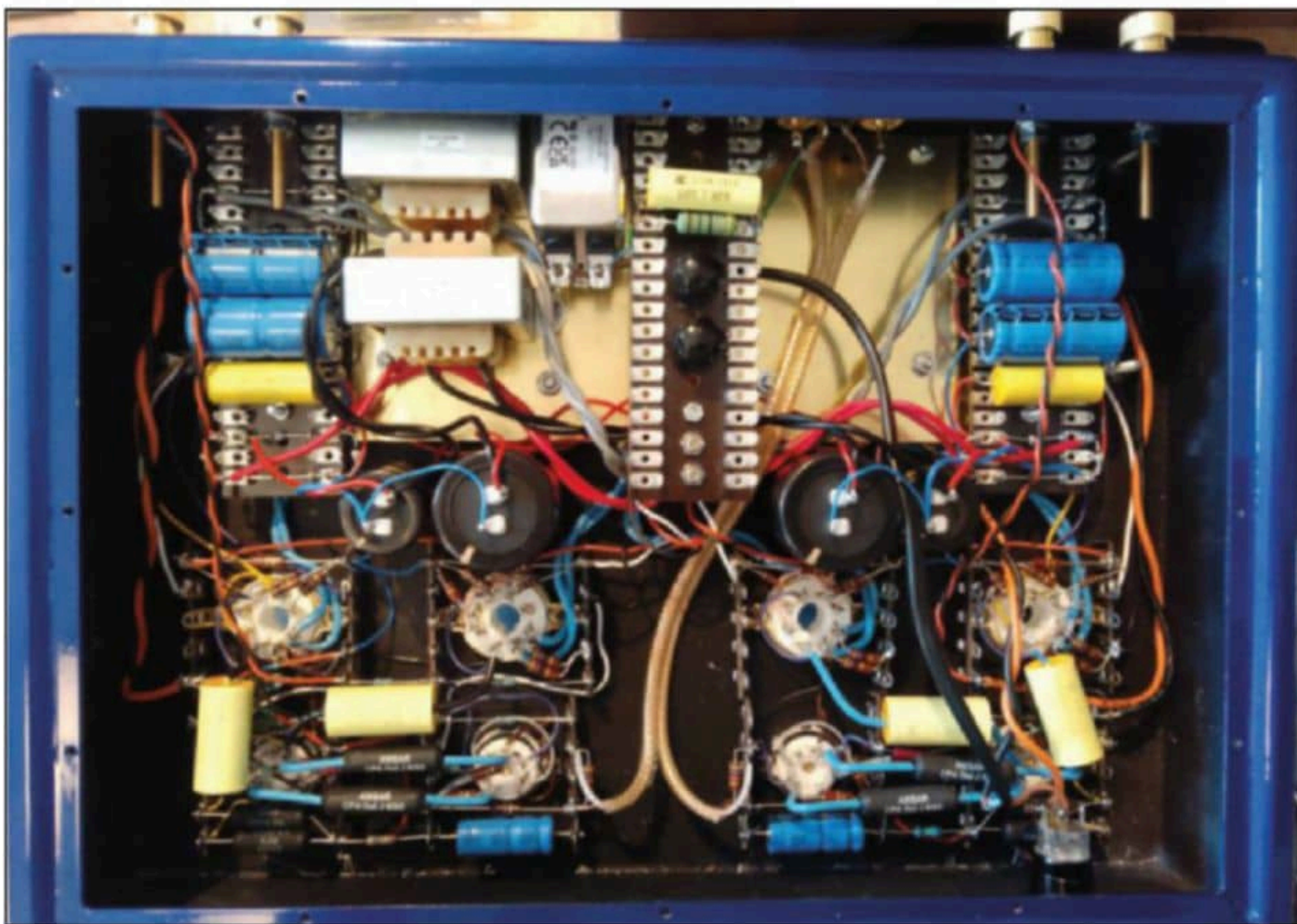


Photo 2: The underside of prototype ZOI amplifier. The channel on the left is wired for a balanced input while that on the right is wired for a single-ended input.

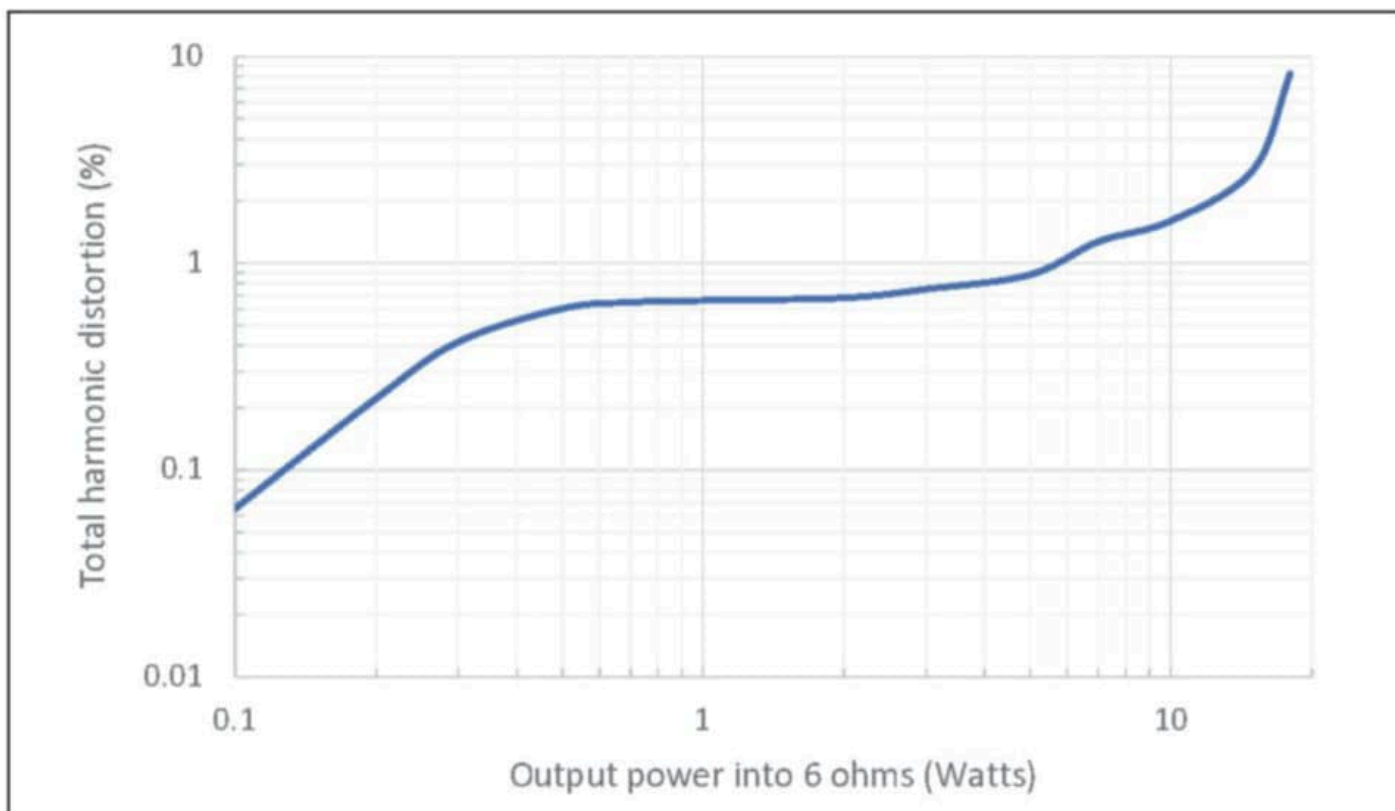


Figure 8: Plot of distortion vs. output power up to 18W.

Performance

In **Figure 8** we see that the distortion rises very gradually with output power, which is typical of a low-feedback triode amplifier with soft clipping. Hence the clipping doesn't sound crunchy like that of a high-feedback design but is more like a form of dynamic compression so that it will subjectively play louder than one would expect from its power rating. At 0.1W, which is some 22dB below the full output of 18W, the distortion is 0.065%. Considering that the typical crest factor of music is 20dB, this is where we do most of our listening at maximum output just below clipping. The soft clipping is illustrated in **Figure 9**.

The frequency response and power bandwidth are highly dependent upon the performance of the output transformer. In the prototype, the output was down 0.8dB at 20kHz with the -3dB roll-off point at 40kHz. However, the small-signal low-frequency response was completely flat as far as could be measured due to the output-impedance cancellation circuit removing the high-pass-filter formed by the transformer primary inductance, although the maximum power fell off below 20Hz due to the falling primary inductance adding to the load as well as core saturation. The zero-output impedance was verified by connecting and disconnecting the 6Ω dummy load while checking there was no discernible difference in output.

Appendix 1: Derivation of Design Formulas for the ZOI Scheme for a Single-Ended Input

Referring to Figure 2 (see Part 1, *audioXpress* May 2024), Kirchhoff's current law states that the sum of the currents flowing into a node is equal to the sum of the currents flowing out of it. Hence, at the node connected to the inverting input we have:

$$I_2 = I_1 + I_3$$

or:

$$\frac{V_0 - V_F}{R_2} = \frac{V_F}{R_1} + \frac{V_F + V_S}{R_3}$$

Rearranging for V_F gives:

$$V_F = \beta V_0 - \alpha V_S \quad (1)$$

where β is the voltage feedback potential divider:

$$\beta = \frac{R_1 R_3}{R_1 R_2 + R_1 R_3 + R_2 R_3}$$

and α is the current feedback potential divider:

$$\alpha = \frac{R_1 R_2}{R_1 R_2 + R_1 R_3 + R_2 R_3}$$

In practice, the values of R_2 and R_3 are much greater than those of R_L and R_S so we can assume that all the output current I_0 flows through R_L and R_S and none of it through R_2 and R_3 . Hence $I_2 \ll I_0$ and $I_3 \ll I_0$, which leads to:

$$V_0 = R_L I_0 \quad (2)$$

and:

$$V_S = R_4 I_0 \quad (3)$$

where:

$$I_0 = \frac{A(V_1 - V_F)}{R_0 + R_L + R_4}$$

Hence, we now have three equations (1), (2), and (3) with three unknowns V_F , V_O , and V_S . Solving yields:

$$\frac{V_0}{V_1} = \frac{G R_L}{R_L + R_C} \quad (4)$$

where G is the no-load closed-loop gain given by:

$$G = \frac{A}{1 + A\beta}$$

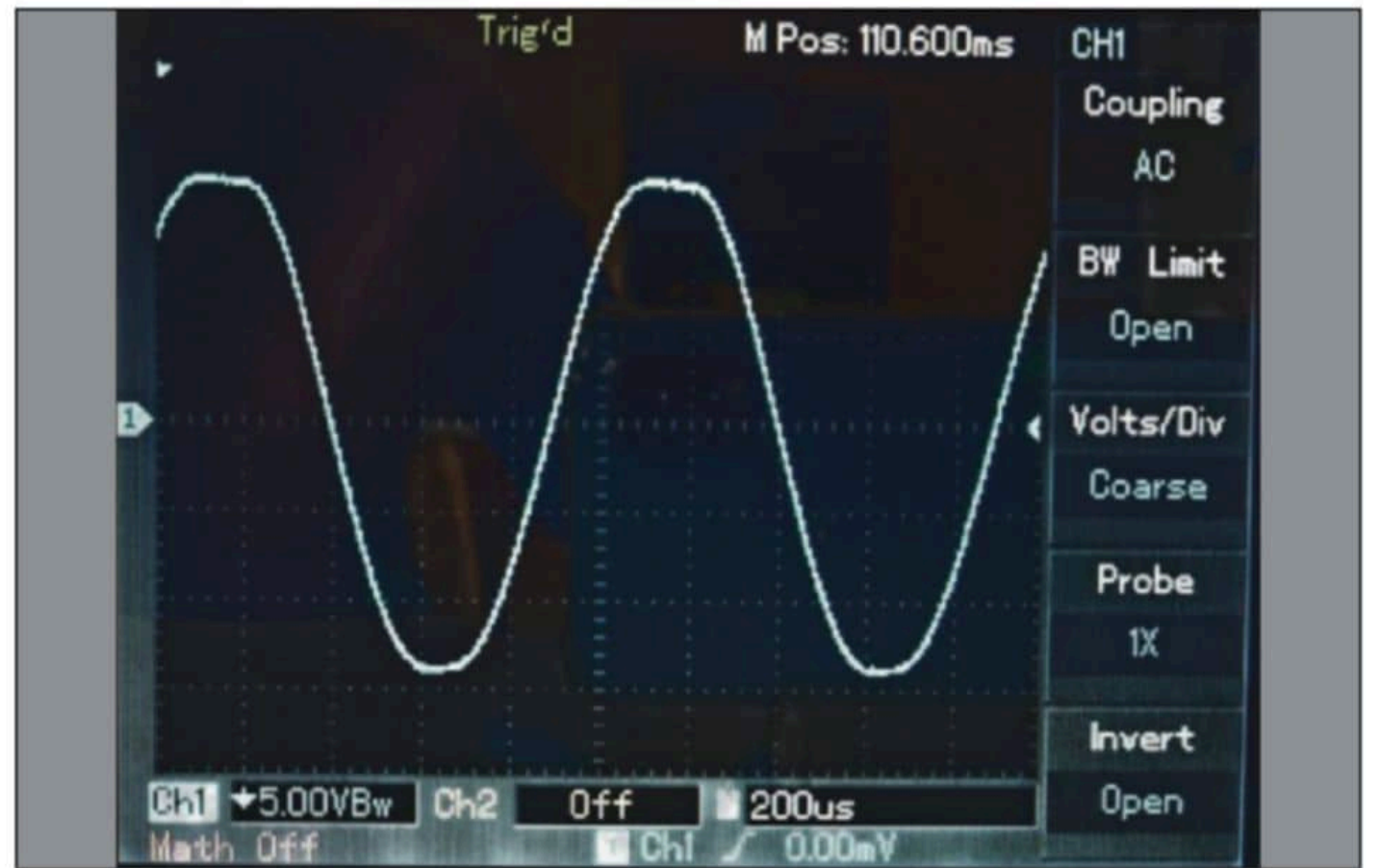


Figure 9: Soft clipping at maximum output.

and R_C is the closed-loop output impedance given by:

$$R_C = \frac{R_0 + (1 - A\alpha)R_4}{1 + A\beta} \quad (5)$$

Let us set the gain of equation (4) to be defined as follows:

$$\frac{V_0}{V_u} = \frac{A R_L}{F(R + R_L + R)} \quad (6)$$

In other words, we set the overall gain to the open-loop loaded gain attenuated by a feedback

Parts List for a Triode Amplifier Using the ZOI Scheme for a Balanced Input

Capacitors

C1, C2, C5, C6 Capacitor, 2.2pF 1% 500V mica
C3, C4 Capacitor, 100nF 10% 630V polypropylene
C7, C8 Capacitor, 470nF 10% 630V polypropylene
C9 Capacitor, 220nF 10% 630V polypropylene
C10 Capacitor, 1μF 5% 400V polyester
C11 Capacitor, 180μF 20% 450V electrolytic
C12 Capacitor, 470μF 20% 450V electrolytic
C13, C14 Capacitor, 47μF -10%+50% 350V electrolytic
C15 Capacitor, 10μF -10%+50% 400V electrolytic

Diodes

D1, D2, D3, D4 Diode (fast recovery), BYV26E 1kV 1A
D5, D6, D7, D8 Diode (fast recovery), UG2D-E3/54 200V 2A

Miscellaneous

FS1 Fuse and holder, 0.5A 20mm
L1 Choke, 1H 240mA
RV1 Trimmer, 50k 15 turn 0.75W + panel-mount adapter
S1 Switch, single pole double throw 20V
S2 Switch, double pole single throw 250V
SK1 Socket, phono
SK2 Socket, XLR
SK3 Terminals, loudspeaker
SK4 Socket, IEC male with filter and 6A fuse

Resistors

R1, R21 Resistor 22k 1% 0.6W 350V metal film
R2, R22 Resistor, 47k 1% 0.6W 350V metal film
R3, R23 Resistor, 48k7 1% 0.6W 350V metal film
R4, R24 Resistor, 0R1 1% 1W wirewound
R5, R9, R14, R25, R29 Resistor, 220k 5% 0.5W 350V carbon
R6, R20, R26 Resistor, 1M 5% 0.5W 350V carbon
R7, R27 Resistor, 22k 5% 5W wirewound
R8, R18, R28 Resistor, 47k 5% 0.5W 350V carbon
R10, R11, R12, R30, R31, R32 Resistor, 100R 5% 0.5W 350V carbon
R13 Resistor, 10R 5% 3W 500V wirewound
R15 Resistor, 4.7K 5% 0.5W 350V carbon
R16 Resistor, 750R 5% 0.5W 350V carbon
R17 PTC
R19 Resistor, 1K 5% 0.5W 350V
R33, R34, R35, R36 Resistor, 47R 5% 0.5W 350V

Transformers

T1 Audio transformer, 25W 3k anode to anode
T2 Mains transformer, 0-180V 0.33A, 0-180V 0.33A, 0-106V 0.03A, 0-106V 0.03A, 0-6.3V 3A, 0-6.3V 3A, 0-6.3V 3A, 0-6.3V 3A

Tubes

V1 Tube, 12AX7 + B9A socket
V2 Tube, 12BH7 + B9A socket
V3, V4 Tube, 6080 + international octal socket

factor F of our choice. Hence, we can define the amount of feedback F as well as the closed-loop output impedance R_C , which can have a negative value. Equating equations (4) and (6) and then solving simultaneously with equation (5) for R_2 and R_3 yields:

$$R_3 = \frac{R_1}{R_4 + R_0 - FR_C} \left((A - F)R_4 - R_0 + FR_C - \frac{(A + 1)R_C^2 + (R_4 + R_0)(R_4 + R_0 - (A + 2)R_C)}{(1 - F)(R_4 + R_0)R_C + (R_4 + R_0 - FR_C)R_L} FR_4 \right)$$

$$R_2 = \left(\frac{A(R_L + R_C)}{F(R_0 + R_L + R_4) - R_L - R_C} - 1 \right) \frac{R_1 R_3}{R_1 + R_3}$$

If we set $R_C = 0$ for ZOI, these simplify to:

$$R_3 = \left(\frac{(A - F)R_4 - R_0}{R_4 + R_0} - \frac{FR_4}{R_L} \right) R_1$$

$$R_2 = \left(\frac{AR_L}{F(R_0 + R_L + R_4) - R_L} - 1 \right) \frac{R_1 R_3}{R_1 + R_3}$$

Appendix 2: Derivation of Design Formulas for the ZOI Scheme for a Balanced Input

Due to symmetry, we assume that the currents in the lower half of the circuit of Figure 5 are equal and opposite to those in the upper half. Furthermore, we assume that the voltage at the non-inverting input terminal is $V_I/2$ while the voltage at the inverting input terminal is $-V_I/2$, both relative to ground. The other three voltages V_F , V_S , and V_O are split in a similar manner. Kirchhoff's current law states that the sum of the currents flowing into a node is equal to the sum of the currents flowing out of it. Hence, at the node connected to the non-inverting input we have:

$$I_1 = I_2 + I_3$$

or:

$$\left(\frac{V_1}{2} - \frac{V_F}{2} \right) \frac{1}{R_1} = \left(\frac{V_F}{2} + \frac{V_0}{2} \right) \frac{1}{R_2} + \left(\frac{V_F}{2} - \frac{V_S}{2} \right) \frac{1}{R_3}$$

Rearranging for V_F gives:

$$V_F = \alpha V_S - \beta V_0 + \gamma V_1 \quad (1)$$

where α is the current feedback potential divider:

$$\alpha = \frac{R_1 R_2}{R_1 R_2 + R_1 R_3 + R_2 R_3}$$

β is the voltage feedback potential divider:

$$\beta = \frac{R_1 R_3}{R_1 R_2 + R_1 R_3 + R_2 R_3}$$

and γ is the input potential divider:

$$\gamma = \frac{R_2 R_3}{R_1 R_2 + R_1 R_3 + R_2 R_3}$$

In practice, the values of R_2 and R_3 are much greater than those of R_L and R_S so we can assume that all the output current

I_0 flows through R_L and R_S and none of it through R_2 and R_3 . Hence $I_2 \ll I_0$ and $I_3 \ll I_0$, which leads to:

$$V_0 = R_L I_0 \quad (2)$$

and:

$$V_S = (R_L + 2R_4) I_0 \quad (3)$$

where:

$$I_0 = \frac{AV_F}{R_0 + 2R_4 + R_L}$$

Hence, we now have three equations (1), (2), and (3) with three unknowns V_F , V_0 , and V_S . Solving yields:

$$\frac{V_0}{V_1} = \frac{GR_L}{R_L + R_C} \quad (4)$$

where G is the no-load closed-loop gain given by:

$$G = \frac{A\gamma}{1 + A(\beta - \alpha)}$$

and R_C is the closed-loop output impedance given by:

$$R_C = \frac{R_0 + 2(1 - A\alpha)R_4}{1 + A(\beta - \alpha)} \quad (5)$$

Let us set the gain of equation (4) to be defined as follows:

$$\frac{V_0}{V_1} = \frac{AR_L}{F(R_0 + 2R_4 + R_L)} \quad (6)$$

In other words, we set the overall gain to the open-loop loaded gain attenuated by a feedback factor F of our choice. Hence, we can define the amount of feedback F as well as the closed-loop output impedance R_C , which can have a negative value. Equating equations (4) and (6) and then solving simultaneously with equation (5) for R_2 and R_3 yields:

$$R_3 = \frac{2A((A-1)R_4 - R_0)R_1(R_L + R_C)}{(F(R_0 + 2R_4 + R_L) + A(R_L + R_C))(R_0 + 2R_4) - (A+1)F(R_0 + 2R_4 + R_L)R_C}$$

$$R_2 = \frac{(A+1)R_1R_3(R_L + R_C)}{F(R_0 + 2R_4 + R_L)R_3 + ((A-1)R_1 - R_3)(R_L + R_C)}$$

If we set $R_C = 0$ for ZOI, these simplify to:

$$R_3 = \frac{2A((A-1)R_4 - R_0)R_1R_L}{(F(R_0 + 2R_4 + R_L) + AR_L)(R_0 + 2R_4)}$$

$$R_2 = \frac{(A+1)R_1R_3R_L}{F(R_0 + 2R_4 + R_L)R_3 + ((A-1)R_1 - R_3)R_L}$$

As a corollary, if we were to use a power op-amp with a differential output, instead of a transformer, in Figure 5 and assume $A = \infty$, equations

(5) and the one before it would become:

$$G = \frac{\gamma}{\beta - \alpha} = \frac{R_2R_3}{(R_3 - R_2)R_1}$$

$$R_C = \frac{\alpha R_4}{\beta - \alpha} = -\frac{R_2R_4}{R_3 - R_2}$$

Solving for R_2 and R_3 gives:

$$R_2 = \frac{GR_1R_4}{R_4 - R_C}$$

$$R_3 = \frac{GR_1R_4}{-R_C}$$

Hence, we could define a negative output impedance R_C . 

About the Authors

Tim Mellow is an acoustical consultant and author who designed the Mellow Acoustics FrontRo hybrid electrostatic loudspeaker, which introduced the "oscillating sphere" concept to broaden the directivity pattern. After 12 years with Nokia, he became a consultant in 2012 and since then has worked with a number of companies on a range of products from headphones, microphone arrays and Bluetooth speakers to ultrasonic transducers for haptic feedback in mid-air. He is the co-author with Leo Beranek of *Acoustics: Sound Fields, Transducers and Vibrations* and has contributed 14 papers to the *Journal of the Acoustical Society of America*. He designed a simple output transformer-less (OTL) valve amplifier with inherent load protection and symmetry throughout, which was published in *audioXpress* in February 2010. Recently, he joined English Acoustics designing circuits and audio transformers.



Peter Farrow graduated from Leeds university in 1989 with a first in electrical and electronic engineering winning the national EEV prize for designing a high-quality digital audio recorder. Upon graduation, he joined the BBC natural history unit in the Special Projects division. While at the BBC he did a further degree in broadcast engineering. He designed the underwater cameras used on the BBC Sea Trek series. He went on to work for Cyrix in the 1990s as a design consultant, designing parts of the M1 6x86 CPU. Peter now owns and runs an Internet Service provider in the UK, a media Production company in Dubai, and is a producer for some high-profile TV shows such as "Will Smith's Bucket List." He started English Acoustics in 2016. Peter has designed many valve amplifiers using an array of valves and topologies, winning awards and recommendations in the hi-fi industry for their sound and build quality.



NAP1 is a time machine.

The Network Audio Processor 1 gives time back to your development team: time to focus on creating future products rather than configuring environments; time to develop features rather than adjusting memory; time to perfect your sound rather than setting up DSP; time to unleash creativity rather than chasing certifications.

Time to reach us to arrange a demonstration

- Single/Dual SHARC+ Cores up to 1 GHz
- ARM A55 Core @ 1 GHz
- DDR3 Memory (8 Gbit)
- SPI Flash Quad (SPI2) Memory (1 Gbit)
- Gigabit Ethernet PHY
- Up to 16x16 channels Dante or AES67 Audio
- Up to 16 channels Dolby Atmos, DTS:X Pro
- Flux capacitor not included



Classic Hollow-State Hi-Fi Before 1950

The term high fidelity (later shortened to “hi-fidelity” and eventually just “hi-fi”) was used as a marketing label in the 1950s to promote the difference in sound reproduction from the technologies available in previous decades. Yet, high fidelity was not exactly a new concept in home music playback systems

By

Richard Honeycutt

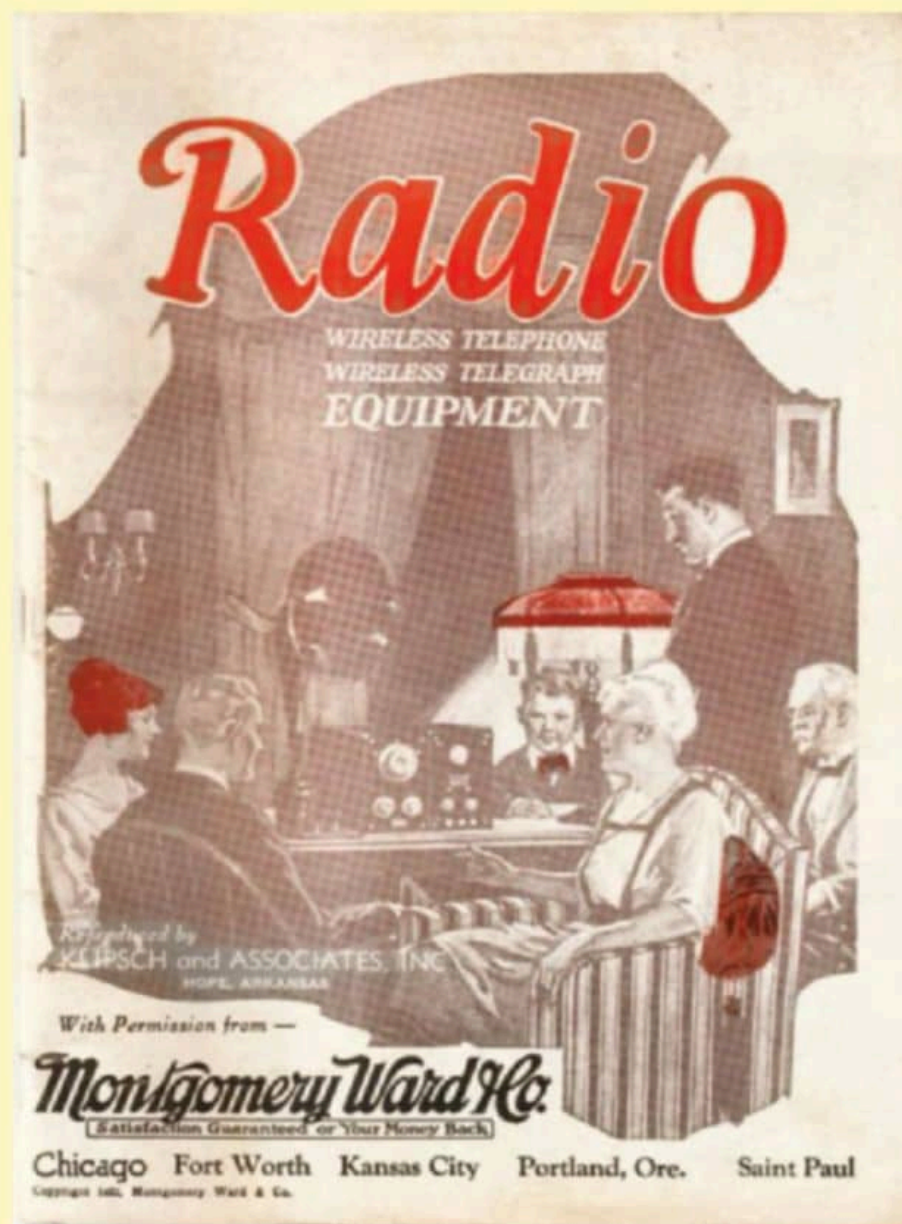


Photo 1: In 1922, Montgomery Ward & Co. published a separate catalog of radio sets, which notably also included radio parts for DIYers!

Although electronic devices for listening to music in the home have been available since the early 20th century, in the form of AM—and later FM—radio receivers (**Photo 1**), and the Edison cylinder—followed by disc, and phonographs, accurate reproduction including broad frequency response and low distortion playback were not possible until the 1930s, and even then, were largely limited to cinemas until after World War II. In the late 1940s, affordable home music playback systems entered the market, and the “high-fidelity” movement was born. Today’s consumer electronics industry is an outgrowth of those early forays into “hi-fi.”

In the US, at least 10 companies were instrumental in the promulgation of the hi-fi movement. Of these, a significant proportion offered at least some of their products in kit form. Let’s review some of the history of these manufacturers.

Knight—Allied Radio Corp.

Allied Radio Corp. was founded in 1928 by Sy Wexler. Its catalog of electronic parts and equipment became the go-to source for many electronics experimenters, DIYers, and later, audiophiles. Knight brand radios appeared in the first (1929) Allied catalog. As shown in **Photo 2**, a Knight radio-phonograph was included in the 1946 catalog. By 1951, the Allied catalog included five pages of hi-fi equipment by manufacturers including Bogen, Masco, Scott, Brook, McIntosh,

Radio Craftsmen, Altec Lansing, Meissner, Bell, and Thordarson, plus a Knight FM/AM tuner 10W and 20W integrated amplifiers. In 1960, the hi-fi-stereo section had expanded to 41 pages and included additional manufacturers, while featuring the Knight KN-740 40W stereo amplifier utilizing two push-pull pairs of 6L6GC output tubes to provide 40W (20W/channel) to 4Ω, 8Ω, or 16Ω speakers (**Photo 3**).

Western Electric’s Offspring Altec Lansing

As most of our readers know, Western Electric (WE) and its research arm, Bell Telephone Laboratories, were responsible for the pioneering development of much of our audio technology. However, most WE audio equipment was built for, and sold to, cinemas. In 1930, WE set up Electrical Research Products, Inc. (ERPI) to serve the cinema market. Due to a 1936 antitrust suit, WE divested ERPI in 1937 and it was reincorporated as Altec Service Co. “Altec” stood for “All Technical,” and the principals of the company agreed to be a service company only, not a manufacturer or seller of audio equipment.

In 1927, James Bullough Lansing and Ken Decker founded the Lansing Manufacturing Co., which produced loudspeakers. Decker died in an airplane crash in 1939, and Lansing sold the company to Altec. The two companies’ names were merged into the Altec Lansing Co. The Model 142B commercial audio power amplifier was introduced that year.

Altec Lansing was purchased by Ling-Temco-Vought, and after 1945, the company began

manufacturing audio electronics for the home hi-fi market. **Photo 4** shows the 1949 Altec Model ALC-101 tube AM-FM tuner.

Fisher Radio Co.

Avery Fisher, a music lover and amateur violinist, founded Philharmonic Radio Co. in partnership with Victor Brociner in 1937. The company manufactured high-fidelity radios: AM sets at first, followed by AM/FM sets. During World War II, production was diverted to armaments.



Knight 5 TUBE PHONO-RADIO COMBINATION

- Powerful Superhet Circuit
- Built-in, Efficient Loop Aerial
- 1½ ounce Crystal Pickup
- Brilliant Tone Quality

TUBES USED: 14Q7 Oscillator-Converter; 14A7 IF Amplifier; 14B6 Detector, A.V.C., First Audio (triple purpose); 50A5 Beam Power Output. Also 35V4 Rectifier.

A well-built, versatile phonograph-radio combination designed for dependable, efficient radio reception and phonograph performance. The handsome cabinet is styled in the functional, modern manner. Its simple lines will harmonize with any interior. Quality components and quality workmanship are combined in the **Knight 5** (with rectifier) to bring you fine phonograph and radio enjoyment for years to come.

Superb Phonograph Performance. Enjoy the finest recorded music faithfully reproduced by this high-quality phonograph. Uses a silent, smooth-running, constant-speed phono motor and a wide-range crystal pickup of the new lightweight type. Needle pressure is only 1½ oz. All 10" and 12" records can be played with the lid closed. Designed for long record life and maximum record enjoyment, the **Knight 5A-163** is the perfect phono-radio for both the lover of classical music and the "boogie-woogie" enthusiast.

Beautiful Radio Reception: A powerful superheterodyne circuit, RCA and Hazeltine licensed, incorporates the latest developments in radio design. Amazing tone quality is achieved—vivid, lifelike reception for full enjoyment. The tuning range is 535-1720 KC. for complete coverage of standard broadcast band and upper police channel. Here are some of the added features: Latest type multi-purpose tubes for greater efficiency; heavy-duty 5-inch PM dynamic speaker for full-bodied tone and extra power handling capacity; automatic volume control circuit to maintain constant volume level for maximum listening pleasure; easy-to-read, illuminated slide-rule dial for smooth tuning; built-in loop antenna for convenience and greater signal strength.

Attractive Cabinet. Made of selected, fine-grained walnut veneer, and styled for simple beauty, the **5A-163** presents a handsome appearance. The cabinet is solidly constructed for vibration-free performance even on heavy symphonic selections. Convenient size and light weight make it easily portable. Dimensions, 14½" wide, 11½" deep, 10½" high. For operation on 110-120 volts, 60 cycles AC. Shpg. wt., 18 lbs.

5A-163 **WRITE FOR PRICE**

Photo 2: The Knight radio-phonograph was a predecessor to a family of hi-fi equipment.

After the war, Fisher sold his interest in Philharmonic and founded Fisher Radio Corp. to manufacture high-fidelity components and equipment. The first products were sold under the trade name "The Fisher." The 50 Pr phono preamp was introduced in the early 1950s (**Photo 5**). The 50A power amplifier (**Photo 6**) followed. Thus, Fisher pioneered the concept of selling hi-fi components as "separates" rather than all-in-one units. Using a push-pull pair of 6S4 triodes as drivers and 1614 "transmitting tubes" operating in Class AB2 as output tubes, the 50A produced an amazing (for a triode amp at the time) 40W output. With dual 5U4GB rectifier tubes and a choke-input power-supply filter, the 50A's power supply voltage regulation was remarkable, resulting in a clean sound at pretty much any power level below clipping.



Photo 3: In 1960 Allied Radio was featuring the Knight KN-740, a 40W stereo integrated amplifier.



Photo 4: The 1949 Altec ALC-101 AM-FM tuner was one of Altec's first entries into the home hi-fi market.



Photo 5: The Fisher 50 Pr phono preamp provided a choice of equalizations since the RIAA standard had not yet been adopted.

In 1957, Fisher introduced the model 500 mono AM/FM receiver using two EL37 output tubes (**Photo 7**). This early mono receiver should not be confused with the later stereo tube receiver models, the 500B and the 500C. These later receivers made in the early 1960s were stereophonic, using push-pull 7591 output tubes. They were also sold with optional wood cabinets and had aluminum faceplates instead of the brass on the earlier 500 receiver.

Fisher Radio has changed owners several times: it was bought by Emerson Electric in 1969, Sanyo Electric in 1975, and Panasonic in 2011. The Fisher brand was replaced by the new owners' name in each of the last two acquisitions.

H. H. Scott

Hermon Hosmer Scott founded H.H. Scott in 1947. The company's first product was the Model 114 Dynaural Noise Suppressor, which was aimed primarily at the professional broadcast market. By suppressing clicks and pops, this unit enabled listeners to enjoy 78 RPM records on their radio receivers, leading to the abandonment of most live radio programming in favor of less expensive recorded music. Some old-time radio personalities described this change as the invention of the "tabletop jukebox."

In 1952, the 10W model 99A integrated amplifier was released (**Photo 8**). This was the first hi-fi amp in which the tubes were

mounted horizontally (**Photo 9**), decreasing the necessary height of audio components.

Scott was highly respected for the quality of its tuners. Scott's chief engineer, Daniel Von Recklinghausen, was certainly a prominent figure in the development and popularization of FM radio. Its first FM-only tuner, the 310A, was introduced in 1954, and the company released its first multiplex adapter in 1961, just after the FCC approved the FM multiplex stereo broadcast standard. Not only did Scott bring to market the first FM multiplex adapter and first FM multiplex tuner, but it also designed and built the first multiplex broadcasting and test equipment. Scott also made kits, and my Scottkit Model LT-111 FM stereo tuner still holds an honored place in my personal stereo system (**Photo 10**).

In 1973, Scott was sold to its European distributor, Electro Audio Dynamics. In 1985, Emerson Electronics bought the H. H. Scott brand.



Photo 6: The 1952 Fisher 50A power amplifier put out 40W.



Photo 7: The first Fisher receiver, the model 500, was introduced in 1957.



Photo 8: The Scott 99-A amplifier introduced the public to the "slim-line" form factor.



Photo 9: This rear view of the 99-A shows the horizontal tube mounting.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.



McIntosh

In 1949, Frank H. McIntosh & Gordon Gow founded McIntosh Labs, which became a legendary manufacturer of hi-fi and stereo equipment. In that year, McIntosh introduced its Unity Coupled Circuit in the 50-watt 50W-1 power amplifier (**Photo 11**). The operation of the Unity Coupled Circuit was discussed in the April 2015 issue of *audioXpress*. This circuit design is still used in the current 75W Model MC75 and the MC275, a stereo hollow-state amplifier. The MC75 was released in 1961, and a stunning Limited-Edition model was offered in 2000 (**Photo 12**). The MC75 used a push-pull pair of KT88s for output tubes, rather than the 6L6Gs used in the 50W-1. Some hi-fi historians date the beginning of the “triode-vs-pentode” disagreement among audiophiles to the preferences stated by audiophiles for either the McIntosh 50W-1 or the Fisher

50A. With less than 1dB difference in output power, one difference suggested to explain the putative difference in the sound of the two rival amps was the pentode output stage of the McIntosh vs. the triode stage of the Fisher. I am not aware of any ABX testing of the two amps, so I cannot comment definitively upon the extent—if any— of actual objective differences in the amps’ sound.



Photo 10: This Scottkit LT-111 FM tuner more than holds its own against any modern FM stereo tuner.

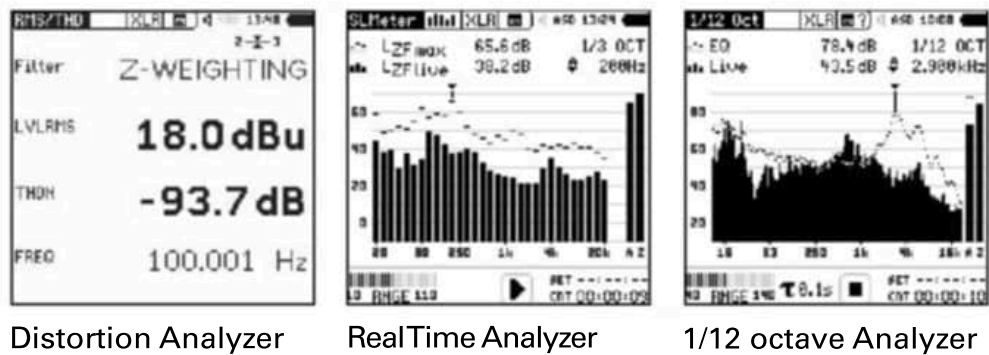


Photo 11: The McIntosh 50W-1 was introduced in 1949.

Ready for any Challenge!

Discover an Audio Analyzer you can trust for your application:

- Sound Level Meter class 1
- Level RMS / THD+N
- 1/1 or 1/3 octave RTA (up to 1/12 optional)



americas@nti-audio.com | (503) 684 7050 | www.nti-audio.com/XL2



Ultimately (by 1969), McIntosh MC3500 hollow-state power amps were the mainstays of the 10,000W sound system used at the Woodstock Festival. Few, if any other companies, still manufacture tube amps of comparable power. Of course, it could be argued that rock festivals do not exactly represent critical listening, so the McIntoshes were used because of power and reliability considerations. Even with the renewed interest in tube amps, accompanied by product offerings from new manufacturers (as well as groundbreaking products from old manufacturers such as Western Electric), McIntosh's introduction of the 130W/channel MC2000 using eight KT88s in 1999 is noteworthy.



Photo 12: The McIntosh MC-75 Limited Edition power amp debuted in the year 2000.

Eico

One of the best-known manufacturers of hi-fi equipment kits, Eico, was founded in 1945 by Harry Ashley as a test equipment manufacturer. The company began making hi-fi components and kits in the early 1950s.

One of the most highly touted Eico audio products is the HF-81 stereo power amplifier (**Photo 13**), which was introduced in 1959 and rated at 14W/channel. It used an ultra-linear circuit and was the recipient of very favorable press reviews.

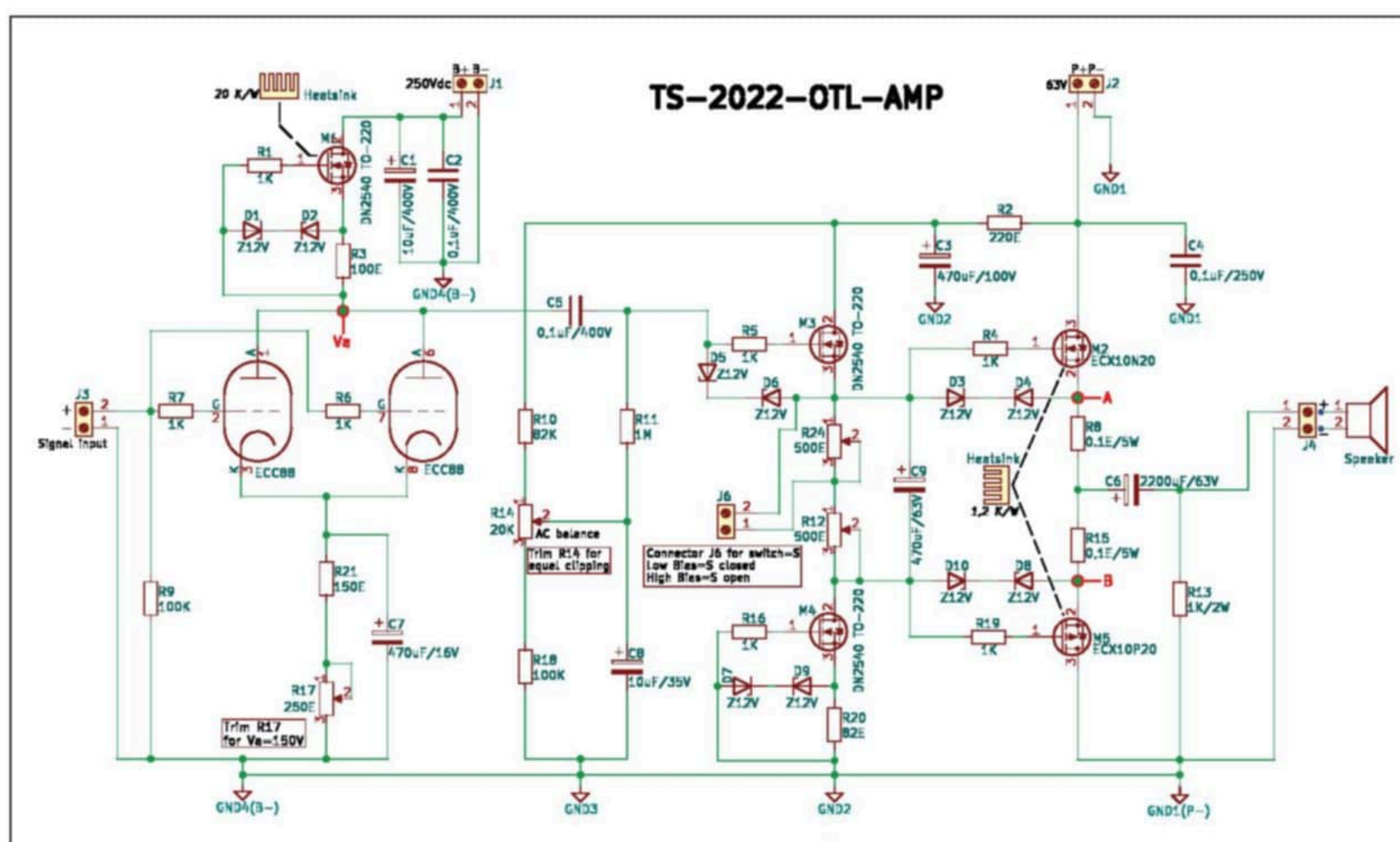
Later...

Some of the most popular hi-fi components were released after 1950. A future Hollow State article will feature the manufacturers and these components. Stay tuned! 



Photo 13: The Eico HF-81 stereo power amp produced 14W/channel from two EL84 push-pull pairs.

In the printed edition of the May 2024 issue of *audioXpress*, the capacitor C5 was inadvertently omitted from Figure 1 on p. 10 of “The TubeSociety 2022 OTL Hybrid Tube Amplifier” article. The figure was corrected in the digital edition. The correct version of Figure 1 is also shown here.



J SERIES

THE LEGENDARY J SERIES IS BACK!



TRIAD'S CLASSIC J SERIES TRANSFORMER DESIGN COMES OUT OF THE VAULT WITH ITS LEGENDARY PERFORMANCE PEDIGREE AND MORE. BORN IN THE 1950S, THE J SERIES TRANSFORMERS FEATURES HIGH-NICKEL LAMINATIONS, PAPER SECTION WINDINGS AND A SPECIALLY TOOLED MU-METAL SHIELDING ENCLOSURE. LONG RECOGNIZED BY ENGINEERS FOR ITS QUALITY SOUND, TODAY'S J SERIES IS IDEAL FOR NEW OR VINTAGE AUDIO GEAR, SOUND PROCESSING AND ANALOG STUDIO RECORDING EQUIPMENT.

SHIELDED AUDIO IMPEDANCE MATCHING
DESIGN FOR A WARM SIGNAL.

SINGLE-HOLE MOUNTING ALLOWS ROTATION
FOR MAXIMUM HUM REDUCTION.

VINTAGE CIRCUITRY, PREMIUM
MATERIALS, HAND-CRAFTED QUALITY.

VISIT ONLINECOMPONENTS.COM TO
ORDER OR LEARN MORE.

MADE IN THE U.S.A.



onlinecomponents.com
THE ONLINE DISTRIBUTOR OF ELECTRONIC COMPONENTS

Modern Design Meets Vintage Sound



DAYTONAUDIO

Dayton Audio Integrated Stereo Hybrid Tube Amplifiers

Featuring the best of modern amp technology and functionality, the Dayton Audio HTA series amplifiers provide all of the convenience you would expect from an amp with the added benefit of classic vacuum tube sonic performance.

- Bluetooth 5.0 for increased distance and lower latency
- USB DAC for direct connection to any PC so you can stream digital lossless audio
- High-quality VU meters for improved visual response and vintage aesthetics (HTA100 and HTA200 only)
- Forward facing headphone amplifier jack for personal listening



▲ The Dayton Audio HTA200 is the biggest and baddest hybrid tube amplifier available. It takes all the best features of the HTA100 and adds more power: 100W per channel. With a digital display, a wireless IR remote and a motorized volume control, the new HTA200 delivers peak class A/B performance for those that seek a unique audio experience.



HTA20
Integrated Stereo Hybrid
Tube Amplifier 20 Watts



HTA100
Integrated Stereo Hybrid
Tube Amplifier 100 Watts



HTA200
Integrated Stereo Hybrid
Tube Amplifier 200 Watts

DAYTONAUDIO
daytonaudio.com

Distributed by:

PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS
parts-express.com 1-800-338-0531
Custom OEM inquiries welcome



To learn more
about the HTA200,
Scan Here!



With a Dayton Audio 5 year warranty, it's more than a guarantee, it's a promise. We stand behind our products and our warranty is a demonstration of our commitment to quality and reliability.